Condemnation without Examination is Prejudice,
or
Words of Wisdom,

by John Curl

May 2006

John Curl at CES 2001

John Curl at CES 2002.

John Curl at CES 2006

“I am always giving stuff away for free. It is not in my best interest, and I am continually reminded of this by my associates. It is so easy! I already know it. Yet, when someone tries to patent your suggestion, without your knowledge, trust me, it is a wakeup call.”

“By the way, I only developed some of the topologies that you folks use, but I am familiar with all of them. Who do you think first developed most of the circuits that you make here on DIY audio? Do you have any idea?”

J. Curl

1 About myself

I am an audio design consultant, have been for more than 30 years, and I design several new or improved products every year. I KNOW HOW to develop the topology and get the basic performance from audio circuits. However, this is NOT enough, IF I want to make something other than a mid-fi product. I have proven this to myself by allowing others, in the past, to make the 'minor decisions' such as connectors, layout, wiring, etc and have paid the price of poor sales, and a diminishment of my design reputation. I could NOT MEASURE any problems, with previous decisions, just lost sales and less enthusiastic reviews.

I did some of my best work when I worked at AMPEX, as well. What a place!
A wonderful technical library. I virtually zeroxed it for the audio info. Still have many obscure zeroxes going back to the '30's. I worked in Instrumentation, Audio, and Research/video, when I was there. We even had a digital audio recorder: 12 bits/50kHz clock. I independently invented the complementary differential input stage and learned low noise design while working there. In 1969, I developed a complementary differential, balanced bridge, 2000W power amp with current controlled output (high Z) for a motor drive application, while working in the Research Dept. Those were the days! I worked at Ampex in the years 1967-1969. I did not work with Watkinson.

33 years ago, I started working with the Grateful Dead. I learned what worked and what didn't work by testing them in live music performance. I tried all kinds of things, transformers, open loop circuits, quality IC's, all kinds of stuff. I learned how to make successful audio designs, that stand up today, sonically, before I ever worked with Mark Levinson. Mark was a good craftsman, but I had to show him the direction to go, which I had previously tested with live music.

I am in the business of serving the audio public. The only 'entertainment' that I provide is being laughed at by others who are not as successful in making audio products. I usually get along fine with my competitors, who are on this website, as we respect each other, even if we do not agree on every detail. What I try to do is to make circuits that give people extra pleasure, much like homemade ice cream or a good tasting wine, compared to the cheapest variety of either. Midfi gives you the 'cost effective' store bought variety.

I was always told that: "Contempt, without examination, is PREJUDICE" I believe that not trying something, not attempting to make actual measurements, and showing contempt of anything outside ones own paradigm is effectively, prejudice

2 Psychoacoustics

Human hearing

In fact, someone could make a sine-square comparison at 20KHz or even 10KHz, and 'prove' that it is inaudible.
Still, I made a test about 20 years ago with three people. I used a Pioneer ribbon tweeter with a measured response of more than 45KHz, with an Electrocompaniet, Otala based power amp, and a function generator. As I remember, I set the function generator with a 5KHz square wave and deliberately limited the risetime to 3.5us, which is about a 100K response, with a quality film polystyrene cap to ground. The function generator had a buffered 50 ohm output. Then, during the test, I added another polystyrene cap in parallel to ground to change the effective risetime to 10us, or 35KHz. We all could hear the difference. It was fairly easy to, as well. Why? I don't know, but we seem to be sensitive to rate-of-change, more than actual frequency response.

I did the best that I could to keep everything in check, BUT there are always tiny differences.

Harmonic perception

I don't know of any definitive info that demands a series of harmonics in a certain way, EXCEPT that higher order odd components are very bad. This includes 7th,9th, etc. However, I agree with the right to state the opinion that the large 2nd harmonics makes a good sounding amp.
In earlier years, I used to go all out and manually balance out any residual even order harmonics, in order to get the distortion as low as possible. Today, I leave a little to a lot of even order harmonics in the final output, because this hypothesis has some merit.

Let me help with this even-odd topic:
'Science and Music' Sir James Jeans 1937 p. 87 This book is available through Dover.
"... The seventh harmonic, however, introduces an element of discord; if the fundamental note is c', its pitch is approximately b (flat) , which forms a dissonace with c. The same is true of the ninth, eleventh, thirteenth, and all higher ODD-numbered harmonics; these add dissonance as well as shrillness to the fundamental tone, and so introduce a roughness or harshness into the composite sound. The resultant quality of tone is often described as METALLIC"

Well folks what do you think that this means?
This is not the first time that I have quoted this passage over the years, but it just gets ignored by those who would not learn from it.

Still, I hope that you can understand that higher order ODD harmonics are sonically problematic. Even harmonics, if extended beyond the 10th or so, might also be a problem as well, but they are difficult to generate in any significant amounts to be very important. It can be debated whether 2'nd harmonic is really necessary to make good sounding audio reproduction. Analog tape, for example, has virtually no 2'nd harmonic distortion, just 3rd and sometimes 5th harmonic, yet can be very pleasant to listen to, without adding 2'nd harmonic to the mix.

Transistor sound

First of all, 3'rd harmonic is actually almost as acceptable as 2'nd harmonic. How do we know? Because 3'rd harmonic is the only distortion normally measurable on analog magnetic tape, and its value is usually between .1% and 10% depending on output level. How is it that we can, or could listen to analog tapes without crying out in pain?
It is the HIGHER ORDER ODD HARMONICS that are the big problem. For example, 7th and 9th harmonic. 8th harmonic should be OK, within reason.
Transistors generate much more higher order distortion than do tubes. This is because of the curvature of their transfer function, due to very nonlinear Gm. Tubes also have some nonlinearity in gain, but they work on a different principle and change less over current, and usually have less distortion, and it will be of lower order in general.

There is a big difference between amplifier bandwidth and high frequency amplifier performance (slew rate, non-linear capacitance, xover distortion, etc). Amplifier bandwidth is usually determined by the amount of negative feedback available. With tubes, this is usually limited by the LOW FREQUENCY oscillation called 'motorboating', not just by high frequency techniques. This usually limits the max feedback in the tube amps to about 20dB.
With IC's and discrete power amps, you can have 80+ dB of feedback, because you can direct couple the stages and have no low frequency problems. Also, you can avoid transformers in the output stage. This gives you bandwidth, but not necessarily better performance at high frequencies.
When you put RF into a tube amp, you should just get a rolloff in level, without slew rate limiting. However, with a solid state power amp, you will usually get slew rate limiting somewhere, or else you have deliberately rolled off the high frequencies at the input. This is an important difference between normal tube designs, and NORMAL solid state designs

I have found that simple is usually better, unless complication can make a better throughpath. Sometimes it does, sometimes it doesn't. Generally, I personally would avoid the complication for my own audio designs.

Distortion compensation

Asymmetric distortion, that is normally spoken of as even order, WILL CANCEL, if you mix it with an inverted signal of the same magnitude. This is normally called push-pull design.
All active components normally create lots of even order distortion, because the voltage gain of all individual active components normally increases with increasing current.
Analog magnetic tape creates [compressive] odd order distortion (S-curve), because away from 0 level, the output loses [gain] in both directions. This cannot be fixed by push-pull operation. However, you can create a different form of 3'rd harmonic [expansive] or inverted S curve from the input signal, and mixing it with the compressive signal distortion, will cancel it out. This is rarely done, because often, the residue created by the process has even more higher order distortion, which is MUCH worse sounding than third harmonic, in almost any amount.

What harmonic does to sound?

It is almost impossible to answer this question. Remember, we are concerned here with sound REPRODUCTION. This implies that we don't want ANY change of the audio harmonics. However, sometimes a little 2'nd harmonic can bring to life a compromised recording. Still, we cannot depend on the need for added distortion to make everything sound its best.

THD limit

It is a waste of time to talk about THD, UNLESS you also know the order of the harmonic(s) that are present. .01% 7th harmonic may be more annoying and even more audible than 0.5% of 2'nd or 3'rd harmonic. This has been known for about 70 years at least. It is just conveniently forgotten about by mid-fi manufacturers.

The facts are that it is easy to make an amp that measures very well at high levels, and poorly at low levels. Just make it class B, which usually generates some crossover distortion. High feedback will tend to hide distortion, but crossover distortion, if serious enough, implies a dead zone, that actually removes gain from the forward path at crossover. This makes things worse than normally expected.
Also, crossover distortion is often hidden in the residual noise (80KHz or so) of the test equipment measurement at low levels, and of course, is reduced in relative amplitude when measuring high signal levels, because the crossover distortion region remains the same level, and the ratio between the two is greatly increased, compared to low signal levels

In order to resolve low level distortion created by crossover distortion artifacts, we usually use some form of spectral analysis on the THD+N residual and reduce the noise floor, perhaps 20-40dB.
Generally, crossover distortion is separate from class B distortion, and rises proportionally at low levels. It is not monotonic with level.

Push-pull class A design cancels even order harmonics, BUT local feedback, even created by another active device in series, will generate odd harmonics from the even harmonics; for example, in a differential pair.
However class B design, such as an output stage, DOES turn even order harmonics into odd order harmonics, because each output device only amplifies 1/2 the sine wave making up the total signal.
Personally, I usually chose to use push-pull class A and differential input stages for solid state. This reduces distortion in general, and makes for DC stable designs. The small amount of extra third generated is not very much, or very important.
However, tubes are another story. Usually, triode tubes can be made so linear that it is a toss-up whether they should be used differentially, except for special applications.
Third harmonic cancellation is another story. It is difficult to do. Still, pure third is not so bad. After all, analog magnetic tape had typically 1% third harmonic at operating level, increasing to over 15% on peaks, yet it could sound pretty darn good. Third harmonic distortion, and its IM products are still close to the music.
It is the 5th, 7th and 9th harmonics that should be removed or avoided at any cost.

Active devices usually naturally produce 'expanding' third, because their transconductance rises with output current.
However, even a small amount of local feedback, often necessary for temperature stability, etc. will convert a portion of the natural second harmonic into 'contracting' third. If you are VERY lucky, you can find a true cancellation, but it usually isn't in a practical circuit.

Subjective sensitivity of human ear

Many of you have NEVER studied the subjective sensitivity of the human ear to distortion artifacts. It was shown about 65 years ago, by German scientists, that you must WEIGHT the harmonic distortion products, IF you want accurate evaluation of the MAGNITUDE of harmonic that is generated.
This was clearly written in the 'Radiotron Designers Handbook' of 1941.
There have been MANY weighting factors: The latest being: N(2)/4 for each harmonic.
You can see that this will give: 2nd=1, 3rd=9/4, 5th=25/4, 7th=49/4 (or than 12 times the 2nd harmonic), 9th=81/4 etc.
I look primarily at 7th harmonic. Why? Because tubes have a difficult time generating it, as well as loudspeakers, BUT solid state can make it easily.
Measure for yourself, if you don't believe me.
Are there other distortions, other than harmonic that elude us? YES!
One is Hirata distortion, shown by Dr. Hirata about 25 years ago, and FM distortion, which like Doppler distortion in loudspeakers, hides in a harmonic measurement. Serious designers must address EVERY type of distortion, in order to make a successful audio product. I certainly do.

I am being rather short with those who can't seem to study simple aspects of audio distortion for themselves, and continually ask the same questions that have been asked for

decades. I realize that some of you are new at this, but try to understand that we addressed these questions decades ago. Yes, we had many of the same questions, but we have found that there remain differences in amplifier designs that still defy our complete understanding.
However, one factor I will just about bet on, and that is the presence of higher order 5th, 7th, 9th harmonic distortion at normal playing levels of almost ANY electronic amp, and you will have a disappointing design. This was stated in 1941 in the 'Radiotron Designers Handbook' as well, but we now allow even LESS distortion these days.
Virtually all Class B designs will generate higher order harmonic distortion. Some will have more, and some less.
Most amplifiers today, are usually high enough in slew rate to not suffer that problem in a severe way, BUT that was not always the case, AND our primary measuring tool in the 50-70's was the IM analyzer, not harmonic distortion measuring equipment. Good oscillators were not yet developed at a reasonable price until the mid '70's, and even then IM was the preferred measurement method. ONLY TIM measurement showed the weakness of IM measurement. I was there when Matti Otala gave our paper at the AES and we were questioned by Crown (politely, I might add) as to why IM did not measure TIM. An analysis of the SMPTE IM test can show that it is insensitive to rate-of-change distortion, but none of us realized it before this time. In fact, the presence of the 7KHz carrier signal seemed to imply that the amp's sensitivity to that frequency would be noted in the measurement, but it wasn't so. Oh well, enough of history.
The main thing is that no one, even today, has every answer to any area of audio design, and there is no one test that will reflect the overall quality of an audio component.

Distortion theory background

Let me give some clues here as to what has happened in audio over the years.
First, about 1/3 of a century ago, my associates and I personally went to visit the late Richard Heyser about the question: "Why is it that loudspeakers, phono cartridges, and analog tape have so much harmonic distortion (lower order of course) that we still hear differences in our electronic amplifier designs, when they can be so low in measured distortion?" Richard Heyser, who was then developing TDS measurement, told us that it appeared that GLOBAL NEGATIVE FEEDBACK was the problem. Could he prove it at the time? No! But experience with negative feedback of his designs and those of others, showed this to be true. Actual harmonic distortion measurements, and even SMPTE IM measurements could be fairly lousy, YET an amp could still sound OK, in many cases. BUT many amplifiers that apparently measured well could sound lousy. Obviously, we had much to learn about distortion measurement.
I went the direction of Otala, in making linear, high open loop, discrete designs.
A few years later, working with Matti Otala and personally making hundreds of measurements, myself, we found that SMPTE IM was most useful for crossover distortion (higher order harmonics at listening level) but useless for TIM (SID) distortion measurement. We were looking for a SINGLE measurement to denote amplifier quality. We tried noise loading, harmonic, SMPTE IM, CCIR IM, sine-square TIM (which we developed), and anything else that we could find.

Ultimately, we could find NO one test would give us complete understanding of an electronic component. This was in 1976, a long time ago.

You people are confused about Otala and his efforts.
First, Matti Otala found, back in the 1960's, by accidently miswiring a power amp, that negative feedback was a problem with the subjective performance of audio circuits. Otala found that when both the open loop bandwidth increased and the feedback was reduced, the amp sounded better.
He then read, as I did at the time, the IEEE paper by Daugherty and Greiner, "Some Design Objectives for Audio Power Amplifiers." 'IEEE Transactions Audio Electroacoustics' pp43-48 Mar 1966.
This article stipulated the QUALITATIVE aspects of TIM distortion, and it was the first to state "... the power amplifier frequency response WITHOUT FEEDBACK that determines the desired pre-amplifier frequency response." This means that the OPEN LOOP RESPONSE has to be higher than 20KHz for optimum operation, according to the D&G article.
Matti then worked on the QUANTITATIVE aspects of TIM distortion with several papers, and he kept the two criteria as important: Low feedback, high open loop bandwidth.
By 1975, many people pointed out that SLEW RATE seemed to be at the heart of TIM. Well it is, except that it is an END CONDITION, like clipping in power amps. It is the behavior, BELOW clipping that is most important, so Matti developed measurements of TIM that occurred below slew rate limiting, and there was plenty of TIM distortion in the 741 type op amps, under almost any reasonable situation. Some individuals found that you could even reduce TIM with an open loop bandwith with only a few hundred cycles or so, if the slew rate was made high and the circuit was fairly linear. Why then, did Matti cling to the 20KHZ open loop bandwidth first put forth by D&G? Well, he found that circuits still sounded better with high open loop bandwidth.
So he put TIM under a more general classification called DIM. DIM contains PIM as well as TIM and maybe other distortion contributions. For the last 25 years it is DIM that we have been working at, the TIM problem being understood.
In 1980, Matti wrote a paper on PIM. This was countered by Bob Cordell,who also tried to minimize the importance of TIM, previously. A personality problem? I think so. So far as I can determine, PIM is not improved by the addition of negative feedback or at least is made worse by reduction in open loop feedback.
I think that PIM is why global negative feedback is still a problem. I could be wrong, but I will design my circuitry to have the highest open loop bandwidth possible, consistent with other requirements.

Walt Jung, independently researched SID (TIM) with extended harmonic distortion measurements. The problem was that existing THD measurement equipment had fairly lousy extended distortion measuring capability, because the IC's used in the test equipment could not remain low distortion at extended frequency to 100KHz, which was necessary. Both Walt and I found that upgrading our ST THD harmonic analyzers with better IC's, that we could make acceptable measurements in order to see TIM (SID, which is the same thing, but it took us years to reconcile this).

Today, the best single measurement approach would probably be a really low distortion harmonic analyzer with discrete distortion harmonic resolution or an FFT spectrum analysis, after the initial THD measurement.

In 1980, Matti Otala, Walt Jung, Marshal Leach and I wrote a complete critique of TIM and related distortions. It wasn't published, but it sure scared Bob Cordell, who was sniping at us at the time, as well as Dr. Cherry. (Do these names sound familiar?) Instead, we published a cover letter 'Audio' July,1980 that saved Bob Cordell the embarrassment of being corrected publicly in print.
Later, Matti wrote a seminal paper on PIM or PHASE INTERMODULATION DISTORTION, that was given, but NOT put into the AES Journal (for political reasons) Since then, Matti has not tried to play in the AES ballpark to any great extent, but went elsewhere.

Still, Bob Cordell, is a minimalist, and worked hard to compromise Dr. Otala at every turn. However, years later, Barrie Gilbert, of Analog Devices, (remember him) rediscovered PIM and gave it a good deal of significance, and a serious problem in op amp type designs.

I am sure that the Cordell article is worth reading, but its conclusions will be essentially negative. What is interesting is that Barrie Gilbert revived this subject approximately 15 years later.

We have had problems with Bob Cordell, previously, with TIM. We wrote a rebuttal to him in 1980, a synopsis of which, was published in 'Audio' in 1980.

A little story about Dr. Cherry. In the late '60's, Cherry and Hooper wrote a textbook on amplifier design. I bought and used the book for years. It had about 1000 pages. NOWHERE, in the book, was any mention of slew rate, 'slope distortion' or rate of change distortion effects.
We found it ironic that in the '80's when Dr Cherry attempted to take over TIM by calling it "slope distortion" Where was he when Dr. Otala was first finding this stuff out? Anyone who is good at math can 'snow' the reader that many factors don't count.
Today, early critics of Dr. Otala, like Barrie Gilbert, have written modern analysis of amp circuits, basically saying the same thing that Dr. Otala set forth in 1980 or earlier. AND, the NEWEST analysis just varifies understanding of the musical listening experience going back to the 1930's. Amplifier designers ignore this understanding at their own peril.

I would like to relate a true story regarding Richard Cabot.
Decades ago, I was at an AES meeting featuring Richard Cabot. He, even then, attacked Otala's work on distortion. He claimed that he did NOT get the same measured results as Otala. However, I got up and asked him WHY he did not use the SAME circuit topologies as Otala, if he expected to get the same results. Don't get me wrong, Richard Cabot is a smart guy. When we meet, on occasion, we are friendly, BUT he had it in for Otala from the get-go. The same goes for Bob Cordell. This does NOT free Otala from any responsibility. Once, after one of his AES lectures, I wanted to get one of Paul

Klipsch's "BULL****" buttons that he handed out at AES conferences and pin it on Matti! Why? Because Matti is a political guy. He did NOT always make it easy to understand his work. In this case, he deliberately avoided relating slew-rate with TIM. This annoyed me as much as the other engineers, but I let it go, because he still contributed so much to understanding audio design.
As far as the name 'TIM' was concerned, it was just as good as 'SID' or 'slope distortion'. It had to be called SOMETHING! And he was first.

Of course, in the '60's, slew rate was a known (but obscure) term. I did a search, many years ago. What Matti was concerned with, was distortion generated WELL BELOW slew-rate limiting. I asked him about this directly, in the 1970's. Every amp and preamp design has a different distortion buildup profile, BELOW actual slew-rate limiting. This is what is most important, and WHY Matti often refused to relate slew-rate limiting to TIM, even though they are related to each other.

Matti had a pretty good gig with Harmon, after they got rid of me. He made plenty of money and got to make an amp.
He still did significant work with the peak current requirement for speakers, IIM, and PIM during and after his Harmon tenure. Let's not sell him short, it was just that he was a born politician. In fact, I am surprised that he did not go into the govenment of Finland in his later years. He was a 'born' congressman, or whatever they elect over there.

The truth is: Barrie Gilbert thought that I was NUTS when I mentioned TIM to him at a ISSCC conference 30 years ago. He had NO idea what I was talking about. LATER, he figured it out and admitted to it. This is normal with traditional circuit designers.

I was in the discussion with Barrie Gilbert, thirty years ago, in 1974. He lives in Oregon, USA. I was born in California, USA. We essentially speak the same language.
Also, I had dinner with Walt Jung for the first time that very evening, and WE discussed both TIM and Matti Otala. Later, in 1976 Matti and I wrote a paper on TIM together. Still later, about 1970, Matti, Marshall Leach, Walt Jung and I got together to write a paper on TIM. This was to head off Bob Cordell, who was chipping at our efforts, in 'Audio' magazine Still later, yet, Dr. Cherry started talking about TIM and wanted to call it 'slope distortion'. And so it goes! It is always easy to find minor faults in the original paths to new understanding. Still, TIM stands today as an important contributor to audio distortion.

Bob Widlar was a design genius, and everybody in my peer group knew it. He got about 1 million $ in stock to move from Fairchild to NS. How about that? His moto was: "I'll drink to that"
If you just replace the compound input pair with a P-fet pair, you can see how it works, easier. This was essentially the UA740.
As far as the detailed writeup is concerned, this is why we go to college.
What is amazing is that the analysis is AFTER the design, by several years. In 1970, I attended a class at UC Berkeley and they did a detailed analysis of the UA741. This was a year or more, after I had used these devices in quantity in servo designs. Sort of an analysis, after the fact.

For the record, I only spoke to Bob Widlar once about audio. (I may have been interviewed by him for a job at Fairchild, in 1963, but I didn't get the job).
This was at an ISSCC conference, in PA, USA in 1974: Bob was on a panel of analog designers. I got up 'knee's quaking' (really) and asked that with the new knowledge of TIM put forth by Matti Otala, where do linear IC's stand with this? He said, that I SHOULD MAKE MY OWN AUDIO OP AMPS, because his designs and those of the others, as well, were NOT designed for audio.
Walt Jung was in the audience, and he came up to me afterward, we went to dinner that night, and have remained friends ever since.

My interest in open loop bandwidth was re-ignited when I read the paper by Barrie Gilbert on PIM as well as other problems with op amp type designs.
When I hear that someone is completely pleased with a 5534 IC op amp, it makes me laugh.
When this design first came out, I knew that it met the TIM criterion, but what would it sound like? Well, I built a test comparator between an IC op amp and one of my discrete designs. It was for a phono stage, and I SWAPPED the RIAA components between the two designs, so that the SAME eq was provided. I spoke about this in 1979 or so, in 'The Audio Amateur' in my discussion with Dr. Lipshitz. I then measured the differences between the two circuits and found them to be very small. However, in listening, the IC op amp circuit sounded too 'smooth', like it was removing detail. I then knew that we had NOT solved the problem with audio problems with negative feedback, just by building a somewhat better IC.
Barrie's paper apparently has been removed from the internet. We might have to contact him directly in order to get a clean copy.
I can't find a copy at the moment, but when Barrie Gilbert actually admits to a problem with op amps, I listen! You should too!

Still, Barrie Gilbert's article shows a unique FM distortion that would NOT be measured by the AM sensitive THD measurement. This FM distortion is NOT fixed by negative feedback, although the AM distortion component generated by the same source has been reduced by negative feedback. This is what makes Barrie Gilbert's analysis significant and useful. It shows a potential distortion that is NOT removed by global negative feedback.

Barrie Gilbert has raised the question and PROVED what Otala had stated on PIM (about 20 years before) and even gave him grudging credit for TIM. I, personally, have make 100s of measurements, both in Dr. Otala's lab in 1976, or more recently with my own, virtually identical test equipment, PROVING the cause and effects of TIM. This is not an issue anymore. PIM is more challenging, BECAUSE we don't have an easy way to measure FM modulation with conventional test equipment. Why not make yourself useful, and show me a convenient way to measure PIM?

The Barrie Gilbert article is called 'Are Op Amps Really Linear?' Analog Avenue www.chipcenter.com/analog/ Nov 28, 2000

PIM and Gilbert paper

Actually the same thing can be shown in a direct radiator loudspeaker. As you know, the loudspeaker must move in and out, in order to produce sound. This causes the frequency of each not played by the loudspeaker to change frequency slightly as the cone is going forward, and then change in the other direction when the cone is going backward.
From what I remember, this distortion can be seen as 1'st order IM (2nd harmonic), but not measured as harmonic distortion.
This is most probably similar to what Barrie Gilbert's article is pointing to.

Either the dynamic phase shift manifests itself as a separate entity, or Barrie Gilbert is barking up a tree. After all, most amps, and IC's can measure down to -100dB or below under reasonable conditions.
Almost all professional audio test equipment can measure to -100dB, and some can measure below this by 20-40dB. You should ask Bruno to confirm your predictions. He has the equipment to do it.

Barrie Gilbert wrote that AFTER 25 years of first learning about Otala's 'discovery' I would hope that you too will learn what is really happening 25 years from now, or even sooner.

Barrie is a very competitive guy, and he thought that I was crazy when I discussed TIM with him in Feb 1974. He didn't give me any real input at the time, except that he had no idea what I was talking about. Now it is easy and obvious.
This is typical of academic types.
They tend to only give begrudging credit to others. In this latest paper, he didn't even footnote Matti's paper on PIM given almost 20 years earlier. Still, better late than never.
A small footnote: Barrie Gilbert holds an 'honorary doctorate' by Oregon State University, but Matti Otala earned his PhD on his own and has been a professor. To refer to him as 'Mister Otala' in this paper is an implied insult.

Well, at least we have had an interesting intellectual 'foodfight'.
Well, this is how I see it, from a historical perspective:
"Those who chose to ignore history, will live to repeat it."
We seem to understand PIM in op amp designs about as well as we understood TIM (30 years ago). What this means is that we are 'stumbling' around trying to understand something that might help us make better audio products.
Let me 'again' give a brief history to TIM to clarify this.
1966, D&G IEEE Trans Audio: Qualitative description of TIM and first recommendation of high open-loop bandwith
1970 MO IEEE Trans Audio: Circuit examples TIM, still qualitative. 1966, D&G article footnoted.
1973 MO & JL at Phillips Research: Power amp design example with low TIM Excellent amp, I bought the prototype and used it for 15 years. Have 2'nd amp of same design in my lab audio system today. Something is 'right sounding' about this amp.

1976 MO, L & JC at Finnish Gvt Lab: Quantitative measurements of TIM and creation of standardized test.
1978 JC at own lab: Measurements of what would cause TIM in real audio sources.
ETC

Where are we with PIM today, compared to TIM? I would say about: 1970.
1980 MO PIM AES paper given in Europe. Qualitative only
1998 BG PIM paper discussed at this time. Quantitative examples.

PIM

The requirement for high open loop bandwidth is NOT a myth.
It is true that you can eliminate TIM, but you can't eliminate PIM without high open loop bandwidth. Matti Otala 'knew' that something was amiss, so he did not back down from the requirement of high open loop bandwidth, even when the slew rate requirement was met. It turns out that he was correct. PIM is as important as TIM. This is why many IC op amps with fairly high slew rates, can still sound relatively lousy.
If you don't believe me, well 'Live in ignorance!'

Distortion

Any competent designer, including Charles Hansen, and Nelson Pass, can make amps that have almost NO measurable distortion. We have found that this is not as important as the amplifier sounding its best. We know this from experience.

There are MANY kinds of distortion. Do any of you skeptics know about FM distortion that caused by excessive negative feedback? I thought not.
Nelson, Charles, Dr. Candy and I are all experienced audio designers, BUT we have different approaches to the final goal.
If you want LOW distortion, buy Halcro. If you want medium distortion, buy Parasound. IF you want do kick back and relax to 'smooth' sound, buy from Charles or Nelson. They trade some harmonic distortion (AM) to entirely remove FM distortion. I compromise somewhere in between, because I have to meet THX specs.

I found one article that gives an overview of the problem: It was written for 'Electronic Design' back in 1998. The link is:
www.elecdesign.com/Articles/ArticleID/7207/7207.html
I hope I got it transcribed right, however this short article references the earlier work by Otala and Barrie Gilbert.
Otala's main contribution is: "Feedback Generated Phase Modulation in Audio Amplifiers" 65th Convention AES 1980, London. Preprint #1576
This is a tough read, and not for the math challenged.
Yes, they seem to have removed access to that Barrie Gilbert article. That specific article is important, because it shows that Matti Otala was on the right track in 1980. Actually, most of you here can get the necessary concept of FM distortion from Walt Jung's article. The 'proof' is fairly math intensive. If you cannot appreciate what is said by Walt's article,

then there is little to further say in order to explain why we use as little negative feedback as possible, in many modern designs.

There have been studies of higher order distortion and its 'annoyance' factor since the 1930's. You could read the 1941 'Radiotron Designers Handbook', like I did, and know that even before WW2, higher order odd distortion was considered very bad for audio quality. Some of the references in the Boyk link would give you even deeper insight.

dB scale

DB is usually used to keep everything on the chart. My equipment can measure either way, so I might look at it more closely. The weighting factor could be converted to decibels, and then added to the graph. It might make interesting comparisons.
Actually, I don't take much stock in absolute harmonic amplitude. We know that global negative feedback can reduce distortion to virtually unmeasurable levels, but we seem to still hear the amp's 'character'. In fact, I have designed two amps that measure almost exactly the same, except for a power increase of about 2 at most , and these two amps sound significatly different from each other. Go figure.

the dB scale is used throughout the engineering community. It is used in RF, instrumentation, and audio. It can confuse things for engineers on some occasions. For example, on a data sheet, transistor Beta change with current is usually expressed in dB. This is a problem, BECAUSE it makes the Beta curve appear TOO FLAT.
By the way, have you seen MY power amplifier curves in 'Stereophile'? Check it out in FEB 2003, or you might look at my earlier design in JAN 2000. I don't have to apologize for ANYTHING!
I use global negative feedback, so my distortion curves are going to look pretty good. Charles does not, so his distortion curves are going to have a higher level of distortion. However, measuring CLIPPING is dumb. Even measurements over 50W for fine detail are suspect.

I know the origin of the dB scale for audio. However, it is used more universally than JUST for audio by engineers. For example, RF. If you think that I should have mentioned it previously, well, I don't teach high school. I presume that people basically understand the difference between the log scale and the linear scale. However, I wish to point out that it is not accurately presented, even in dB, UNLESS a weighting factor is added and then the most important harmonics noted.
Also, I wish to point out that class A linear amplifier stages will increase in distortion as power output is increased, and visa-versa. The rate of change of this increase is related to the order of the harmonic. Therefore, some 7th harmonic, for example at 100W, would be MUCH LOWER in level at10W or so.
This makes high power spectrum measurements misleading as to the relevance of the amount of higher order products seen on the plot.

At very high levels, all class A amps are going to generate their worst distortion. Only extremely high feedback can suppress this. This is why I use global feedback. I have to meet THX specs. This is important for Parasound, and our dealers.
I personally think that zero global feedback is usually best, all else being equal. I noticed that the 'Stereophile' graphs use 20dB/division. This is because the fundamental is not nulled. I generally use 10db/division in my measurements. It changes the 'compression' of the graph. I also somewhat suspicious of the residual higher order distortion present in the test signal. Even the 'exceptions' in the higher order distortion could be cancellation of the residual distortion by the amp. I can't be sure.

I referenced measurements of two power amps that I have been associated with on this thread, the 3500 (350W) and the JC-1 (400W). If you look at these measurements closely, it is difficult, if not impossible to predict the actual audio quality from the measurements. They are virtually the same, except for somewhat higher power output in the JC-1, BUT then it costs almost 3 times as much as well. These amps, with almost the same schematic, DON'T sound the same. I believe this comes from better layout and parts selection, not the basic design. I wish that I could find a way to measure why. Incidently, earlier models, not as sophisticated in design, actually sounded better than the HCA-3500. This is where success can skip model generations.

We have tried a number of different tests, including: Two, 3, and multitone IM, noise loading, and other tests. These are sensitive tests for static distortion, better than harmonic distortion, BUT difficult to get extremely high resolution.
There is one important component that we have not completely addressed. This is FM distortion, due to the working amplifier bandwidth being modulated by dynamic changes in open loop gain, due to open loop distortion. The only reasonable cures for this, at this time, appear to be high open loop bandwidth and moderate feedback, or little or no global feedback. Duh!
This has been called IIM distortion by Matti Otala about 20 years ago, and fairly recently was considered as an important distortion factor by Barrie Gilbert of Analog Devices.

2.1 Listening tests

Listening tests

My listening tests, using the best equipment available at the time, often gives me insight. I use this insight to design better audio equipment. This gives me A ratings in listening contests from audio reviewers. This is what I do for a living.

It is unfortunate that you give the human ear-brain combination so little credit. I find it useful to pay attention to details, including power cords. I have heard differences both in my personal system, and in very expensive audio systems owned by friends and associates. Personally, I wish there were no differences in power cords. I have to design the rest of the electronics in the audio system, and have enough to concern myself with, without including power cords, BUT hearing is believing.

My situation may be different from many others. You see, I actually have to produce successful audio products, not just claim that they are essentially the same (under blind conditions of course), or that we practically know everything from undergraduate college physics/engineering to optimize any audio design.
I prefer to stay ahead of the mid-fi pack.

Blind tests

Is there no difference between 'New Coke' and 'Old Coke'? I think so, BUT a double blind test can be constructed to prove me wrong. Is this science?

Can't everyone see that if a test obscures noting a difference, then it will come out 50:50. Of course, by chance, preferences 'could' be 50:50, but how often would that happen, if real differences were noted? Not very often, I suspect.
almost anyone can do BLIND TESTING, it is ABX testing that is virtually impossible to pass, because it FORCES a decision of a certain kind.

Importance of listening tests

> …few designers trust their ears more than their measurement equipment.

Of course Nelson is correct. This makes a real problem for 'engineers'. They want to put a number on everything. Yet, it is well known that total harmonic distortion (THD) is a LOUSY indicator of audio quality. For those of you who don't know this, it is because TOTAL harmonic averaging mixes the nasty higher order products with the relatively benign 2nd and 3rd harmonic distortion. Yes, 3rd harmonic distortion is relatively benign. If it wasn't, we could never enjoy analog tapes or any music recorded BEFORE digital recording was imposed.
Most of what serious high end designers concern themselves with is beyond normal measurement. This hasn't kept me from investing a lot into test equipment. All else being equal, I prefer the lowest distortion possible, especially reduction of higher order distortion to VERY low levels. Once again, I can only do so much with test equipment, in fact, I have proven to myself that my designs can sound lousy, yet measure just fine, yet with 'refinement' of the components in the circuit, my associates and I can make an outstanding product with the same topology. Why? I don't know, but I do it on a consistent basis. Only our ears can tell the difference, under these conditions. This is the 'extra' factor that designers can add to audio design, above and beyond topology improvements. It is difficult to put this into a book, although it deserves serious consideration.

Hearing impaired

Older people hear pretty well, IF they are audiophiles. Actually the 20-20KHz is sort of an average. I could hear 24KHz when I was young, and many 'ultrasonic' alarms annoyed me. Even as we get older, we can hear the IM distortion in the midrange, and the compromise of high frequency by CD and other sources. Generally, most loudspeakers do not really go much beyond 20KHz, if that far. This is another real problem. Now, supertweeters are being developed to fill in, and for some people, and some new sources, it might be useful.

Blind tests

For example, about 25 years ago while in Japan for HK, I was asked to listen to 3 separate audio circuits, not made by me, in a blind test. I could distinguish them and point out which was the best. It amazed the Japanese, but it did not surprise me. I do it all the time.

If you have 2 or 3 selections and that you taste each one, and then you are given an unmarked selection and you decide which of the original selections that it is. You do this 20 or more times in a row, and you have to be right 95% of the time to have any significance to your decisions. Wow! I once observed a bar bet like this, and the person couldn't tell the difference between cola, 7up or ginger ale after a few tries. Of course, to be fair, we must make the wines a similar as possible. I might suggest adding sugar or other components in order to 'even up' the wines, so that we are not considering taster preference. ;-) Can you see a parallel to this in audio testing?

I think that there are serious differences in the 'wine tasting' test and the ABX test. I think that it really makes a difference in the results, even if the same statistics are employed. It may also be that we are looking for different things, such as 'quality difference' in wines, but we expect the hi fi stuff to be virtually the same 'transmission' of audio information, and if there is a difference, we will first equalize it out, as best we can.

Lipshitz and VanderKooy? I found SERIOUS problems in their tests 25 years ago, and put them in print. Is this the sort of test procedure that you think is realistic?

I do have virtually all the info from the AES on CD rom and all articles from 'The Audio Amateur' where most of the important debate occurred. My basic complaint about the original Lipshitz-Vanderkooy articles in 'TAA ' was the lack to technical understanding to make the testing somewhat on a level playing field. For example, they rolled off the highs 6dB at 15KHz on both units being tested, and apparently didn't notice or care. They also tried to test for slew rate with a moving magnet cartridge that had a 4th order rolloff filter at approximately 20KHz. DUH?
To me it there were other factors similar to wine tasting with equally dirty glasses, rather than clean ones.
My problem with ABX testing itself, was first stated 25 years ago in 'TAA'. This was followed up by subsequent articles in 'TAA' by Rod Rees, who was a professor at Washington State Univeristy (I'm pretty sure) It is all here, if you are interested. You are free to put any info that I give you on this subject to associates or on line here.

As a proud member of the high end audio design team, I think that this obsession with double blind testing shows a problem in itself. It is not really necessary to make a successful audio product. However, if the problems of ABX testing were easy to fix, or the proponents were willing to address many of the criticisms of the test, then it would be used. At this point, it is essentially a NULL test that implies that we are generally wasting our time to attempt to improve audio design.

"You can't fool ALL the people, ALL the time!"
We ALL know that we can sometimes fool ourselves, and others. BUT, that does NOT mean that we would bother to fool ourselves, when we could make something cheaper and more profitable to us, by fooling ourselves and others. We want to make better audio products, this keeps us on our toes to watch out against fooling ourselves in some way.
PS on rereading, I hope that I have not been 'foolish' in the way that I stated this

I make audio products that don't cost much, as well as expensive products. IF, I could make them sound the same, I would, but I can't. Maybe you or someone else can?

I don't quite know where this is going but: I design products or at least assist others in product design at ALL price levels. For example, I might review a $100 retail phono stage and offer suggestions to improve it. At the same time I make a $5000 retail phono stage, that really sounds a lot better. I apply all that I can from my experience with the $5000 phono stage that I can to make the $100 as good as possible, but I'm afraid that it doesn't sound quite as good. If I subjected these 2 preamps to an ABX test, it is quite probable that almost nobody would be able to hear the difference to a 95% statistical level. Does this mean that the $100 unit, used within its capabilities, is just as good sounding as the $5000 unit? If so, then save your money, folks! However, in normal listening tests, there is a significant difference between the phono stages, so I think that there is something wrong with the ABX test method, rather than the two phono stages actually perform at the same quality level.

AB Tests

If I had to prove every design technique that I use by a blind test first, I would never make any progress, and my competitors would evolve past me by trying different things, without regard to the 'scientific method' or somesuch, as they have, often enough before. When I bring up the background of some designer, it is not just that they are qualified, and educated, but that they actually can teach me a few things when I do communicate with them. What have I learned from you? That I am not the same person that I was 30 years ago, when I developed the JC-1 ,2, and 3? Of course I am, except that I use my physics background more these days, compared with the past.

much of the 'mental position' represented here on this thread is shown in books by Robert Alton Wilson. I get much insight from them.

2.2 ABX tests

ABX listening tests

this is my opinion about ABX testing. At present, and probably in the future, it is a waste of time. You can't beat it! It will almost always tell you that two devices sound the same. If you believe this, then get another hobby. You already have good enough electronics, at least for your needs. The problem for me is that I make audio designs for a living. I started off with some darn good circuit topologies, better than many discussed here, 30 years ago. Over time, I have learned to 'improve' these topologies by using better parts, layout, and 'X' factors that I would be laughed at, or at least, hassled by many on this website. I usually do not invent these 'X' factors. I work with others who have a knack for discovering them. Deep down, I really like to measure and understand why things sound like they do, but sometimes I just have to use something without really understanding why, because it sounds better to me, and my associates, than something else or another approach.
This is my position, I use what works for me and the audio public, who use my designs. I don't have to 'prove' anything beyond that.

ABX testing is NOT the standard for serious evaluation of small audio differences.
JJ is a bright guy, BUT he never told us how he did his BLIND TESTS, as it was a company secret, AND he made derisive comments about the ABX tests as they are typically done, over the years.
Our own efforts in evaluating ABX testing showed that it was worthless, like the test of New Coke and Old Coke on the guy who invested 10's of thousands of dollars in a campaign to get 'Old Coke' back to the public. He failed it, even though they brought 'Old Coke' back as 'Coke Classic'. This was just a different disguise of the fact that many people preferred the older formulation, over the newer formulation, even if 'double blind' testing showed that they could NOT tell the difference. Go figure!

Do you know Floyd Toole? Do you work with him?
Most of you have little practical understanding of ABX testing. If you did, you would know that between properly equalized electronics, it is virtually impossible to hear any differences, BECAUSE of the test requirements (limitations). Since I design amps and preamps, I ignore ABX testing, but wish anyone else, who might be my competitor, to embrace it with all enthusiasm!

Folks, just for completeness, I might mention that I have conversed with Floyd Tool, Drs. Lipshitz and Vanderkooy, David Clark (builder of the ABX box) and Les Leventhal on this subject over the years.
It has been shown that the worst caps (tantalum) that we could find, could not be detected in an ABX double-blind test, by one or two of the persons mentioned above. If the worst caps that we could find can't be detected, what is the point? Settle for an IC 150 and listen to music.

I don't know where you get your info about ABX testing, but it is virtually worthless for amps and preamps, UNLESS the devices under test significantly change the amplitude response, or have gross distortion. I generally don't design amps and preamps with gross distortion, or low damping factor, so it is worthless for me to use ABX to try to find any differences.

EVEN IF I could find a difference with an ABX test, it is one of the most insensitive ways to find anything. This is partly because of Les Leventhal's comparison of type 1 and type 2 sensitivity to differences with ABX.

AB or ABC testing works OK. But I hope that everyone understands that we designers were very interested in ABX testing when it first came out. We just found that it did not give us much real input. Just that similar electronics sounded the same, yet they did not sound the same when listened to without ABX testing. Heck, if I believed my own ABX test, I would still be listening to a Dyna PAS-3X preamp, at least for line livel inputs It sounded the same as the Levinson JC-2 to me. In fact, many designers dropped out of competing in the marketplace in the late '70's because they BELIEVED their own ABX test! What was the point of trying to 'improve' anything?

Folks, there are those of you who have never read about the problems with ABX testing or participated in an ABX test yourself. Unfortunately, you just can't make an 'informed opinion', only a wishful one. Study up, and we might have a more enlightened discussion on this subject. I recommend the back issues of 'The Audio Amateur' from 1978-1982+ for the most balanced discussions on the subject. The more fortunate might have a discussion with Les Leventhal on the statistics and how they are stacked against hearing any differences. Those with an engineering bent, might look for serious problems in many ABX tests, such as distortion overlap from sources being paralleled and little attention to 'accuracy' only 'sameness' in the audio signal. For example, if the music used was 6dB down at 15KHz compared to 1KHz, this was considered ok for ABX testing, so long as both components had the same defect in accuracy.

First, I feel that many of you have limited experience and knowledge to the accomplishments that we have made over the decades in audio design. Also, it is important that one actually has discussed ABX testing with the principal promoters of the test, in order to get as much understanding as possible about it, before deciding whether to use it or reject it. Many of you, who criticize me, don't seem to have 'walked the walked the walk, or talked the talk' that is necessary to have an informed opinion on the subject.

Folks, I must apologize when I 'brag' about my past published work, or that of others. It is easier to refer to it directly, rather than to give you an obscure reference, that most of you could never easily find, even if you wanted to bother.

In this case, it is important that I have done research on capacitor distortion and have published it, especially with regards to tantalum coupling caps.

For that degreed mechanical engineer who is so sure of his knowledge:

From 'The Experts Speak' once more: "I can accept the theory of relativity as little as I can accept the existence of atoms and other such dogma' Ernst Mach (Professor of Physics at the University of Vienna) 1913" p299

How about that? What a guy! Still, I have his textbook on my bookshelf.

Although I applaud anyone who has the drive to get a degree in engineering or physics, please learn your limitations. You will learn this from experience, soon enough.

First, I have been in this business a long time, about 40 years in the serious study of audio. My colleagues are and were: Paul Klipsch, Richard Heyser, Matti Otala, Walt Jung, Michael Gerzon, John Meyer, Dick Marsh, Dr Hawksford, etc, etc. This means that have I talked extensively with, shared dinner with, drank with, visited, and was visited by, worked on research with, wrote papers, and was continually on the phone with, etc, etc. This has nothing to do with photo ops, or a polite handshake. We still meet together and talk on the phone when convenient.
These people I have learned from, and they from me.
Michael Gerzon was at least as far out as anything that you have ever heard from me. He, wisely, just didn't put it into print. IF you don't believe me, ask Dr. Peter Craven, his associate. Peter Craven is still with us, and has recently put some important info, that I'm sure he developed with Michael Gerzon in a recent AES journal.
The uncertainy of frequency vs time measurement was first told to me by the late Richard Heyser in the late '60's. Far out stuff, then. It was refreshing for me to see it in Michael Gerzon's article, last night and helps me better understand Heisenberg's explanation in quantum theory.

My audio world is one of growth and experimentation. If I can learn from someone else, I will, therefore I cultivate relationships with interesting people in order to learn and grow. Many of you don't seem to have the same experience. This is unfortunate. What I could say that might be useful is lost in the adversity shown to me. Teach me something, rather than tell me what is impossible, because of your opinion on the subject.

This is like going back 20 years and hearing it all over again. As I remember, CD was "Perfect sound forever!". Dr. Doi's book 'Digital Audio Technology' 1983 should have set us straight. It is virtually all there! Why bother listening, or 'improving' perfect sound already?
Unfortunately, for me, it didn't sound 'perfect' far inferior to a record, analog tape, or even an FM tuner broadcast.
Then, among other things, we discovered 'jitter'. But, how to measure it, and why should such a small change in the time domain of micro, nano, or even picroseconds make any difference? Impossible? Well, I don't think so. In recent years, primarily do to the efforts of hi end designers like Meitner and Bob Stuart, we have seen CD reproduction 'improve' beyond 'Perfect sound forever'.
I guess that these improvements were there all along

In my day, we had Wow and Flutter, this was a problem with analog recording and a real problem with FM recording. I'm sure the military-industrial complex and the telephone company had lots to say on the subject, BEFORE we got interested in it.

This is from 'Digital Audio Technology' Dr. Doi: p144" ... Delta T is the jitter margin, and if jitter exeeding this value occurs, random errors will increase to the level that it becomes impossible for a machine to read the signal correctly. Of course, the reproduction waveform will, as a rule, deviate from the ideal and this in itself will reduce the jitter margin and lead to an increase in code errors. " This basically tells me that so long as the jitter is does not exceed a very large amount, it is unimportant. Is this true?

How about other effects? This was all that was said by Dr. Doi of Sony in 1979. Check it out for yourself.

I just read a 'Positive Feedback' article of an interview with Ed Meitner.
This guy knows his stuff, and I learned a lot from him as well as independently agreed with him on many opinions. What a guy! I wish that I knew him better. I would be proud to associate myself with him.

But for the record, I consider the prerequisites of education, experience, innovation, and success as important factors in understanding someone and their opinions on a subject. Meitner for example, fulfills these catagories successfully, yet even he has been criticized on this thread. What did he do to deserve this? Think about and improve audio products? Is that a valid criticism?

AES

The AES isn't what it used to be. It is almost impossible to get anything of value to design engineers into the AES Journal, these days. This was not always the case, and I recommend anyone to read AES journals 20 or more years old to answer MANY of the questions asked on this website. The latest journals will please some of my critics, but they rarely give me much engineering insight. This is because the AES Journal was taken over, about 25 years ago by Dr. Lipshitz et al, and he made sure that Walt Jung, Matti Otala, and probably anyone else like us, could not easily contribute. We can still give papers at conventions, but that's it. The argument was given that the AES is a 'Journal of Record' which precludes any controversial 'new' material.
This is why 'hi end' has broken away from the AES. There are still some contributors to the AES who have not been completely turned away, such as Dr Hawksford and Dr. Peter Craven. I always look forward to their articles in the AES Jounal, as I know I will get something from them.
PS, I personally have been an active member of the AES since 1966, and have contributed my time to promoting its affairs, just not lately.

the AES is primarily a club for audio business. It is controlled by the likes of Dolby and Lipshitz. They just don't think much of hi end. They prefer mass consumption, like fast food. The AES did not move forward, but in another direction from its initial conception. Personally, I am all for super digital, Class D, and every new thing that can possibly come out in electronics, such as buckytube transistors, etc. Amazingly, it is the younger people who don't have their minds as open to new possibilities as many older people.
I also feel that many younger people have not been exposed to good analog reproduction, so they think digital is the normal scheme of things, much like kids who have only eaten Macdonald's hamburgers, and nothing homemade.
So long as many in the audio industry have low expectations of what is possible, we won't see much progress, except for cost or space savings. Why should they bother to do it better?

3 Feedback

Feedback

We designers know that there are two kinds of feedback: local, and loop (or global) feedback.
To save time, we usually refer to loop (or global) feedback as applying feedback to an amplifier.
This was the principle of negative feedback developed by Black, at Bell labs, in the 1930's. For many years, we thought that global feedback was OK or even the greatest thing since canned milk! However, Otala and others showed us that negative loop feedback caused as many problems as it eliminated. Therefore, many designers, including Otala, Hansen, and Pass, and even me, on occasion, have relied much more or even completely on local feedback, which still presents SOME problems, but not nearly as many sonic problems as loop feedback.
This definition of 'feedback' is much like what we have settled on for 'transistors'. Apparently, 'fets' are "unipolar transistors", and what we call 'transistors' are really" bipolar transistors". We just say fet or transistor, today and we know what we mean. It saves time and trouble.

Feedback

We must differentiate between local feedback, followers, and loop or global feedback. While they have some similarities, they also have significant differences.
It is interesting to look at an ideal follower with R(load) being very high. In this case you have TOTAL degeneration, with the interesting result of the follower being close to ideal, because there is virtually no significant change in Gm, therefore no distortion is generated. Yet, a follower with an 8 ohm load would behave much like a common emitter (source, cathode) with 8 ohms of resistive degeneration, so far as distortion is concerned. In this case, there is almost no 'feedback', but also no voltage gain.

For the record, the reason that a very lightly loaded follower has static Gm is because the I(q) that sets the Gm does NOT appreciably change over either the input or output output swing.

BUT there are different types of feedback. One is series, another parallel, and a third group is global, which can be either series or parallel.
There are lessons to be learned from this:
For example, local feedback gone wild, can be called a follower. This is because, in order to get any meaningful output from a device with a strongly degenerated Gm, you have to use it as a follower. However, in principle, if you put a cascode stage on top, and then an active load in the collector, you could, in theory, have an extremely linear voltage gain stage. Noisy yes, hi Z output, yes, but real voltage gain, and real linearity too!
This implies that a follower is really just a 'series feedback' stage. However it will behave differently than a parallel feedback stage or a global feedback connection. This is what

we want to keep clear. It is not necessarily 'feedback' that is the problem, but how it is implemented. I realize that many will insist that we should not have separated followers from fed-back common emitter pairs, because they use some similar basic mechanism, but they do not behave exactly in the same way. This is important to serious audio designers.

I would say that feedback is sometimes necessary, but I would prefer not to use it, if possible. I feel the same way about coffee.

I would prefer not to use feedback, BUT I can make an acceptable audio product with feedback, if it is necessary to meet mid-fi specs as well. Charles Hansen knows this and has stated it here.
Now who is MY competition? Well, Charles for one, Nelson Pass for another. YES, we really compete with each other in listening contests like the CES, for audio reviews, and in the audio marketplace. I think that we are pretty good sports about it as well. We will still talk to each other and do not publicly 'badmouth' each other, even if we don't necessarily agree on everything. This is called professionalism, folks, or something like it!

Actually, we could, in principle, make a 'zero feedback' amp, if we used power Vfets (not Vmos). They have low output impedance on their drains, at least low enough for a horn loaded loudspeaker. Lots in parallel would work with direct radiators as well.

Feedback and PIM

Actually the feedback question is more complex than expressed here. There is also higher order distortion generated from lower order linearities and FM distortion generated by non-linear input stages within a feedback loop with low open loop bandwidth. Real audio designers take negative feedback very seriously.

Feedback also causes FM distortion. Get out of that one, if you can!

I went through 'Are OP Amps Really Linear?' by Barrie Gilbert who knows his stuff. If you can explain to me where he went wrong when he stated, after much math, that: "Then, the actual phase angle is 6.01 degrees. By E=5, it has increased to 9.14 degrees. This is not what we would expect from a LINEAR amplifier, whose phase should quite independent of amplitude. In certain applications, this excess phase and its variation with signal level will be very troublesome. " Page 6 of 7

In other words, we have a dynamic phase shift that changes with output voltage level, of audio frequencies. I don't find any way that adding feedback fixes this problem.

Feedback and opamps

Most of us use global negative feedback. Although sometimes, we experiment with open loop circuits.

Negative feedback may be necessary and even an improvement to a successful circuit, BUT it is not always so. IF we can reduce or remove global negative feedback, and still maintain fairly low distortion, we usually find that the open circuit sounds better. WHY? That is the question that we have been addressing for decades. Is is phase modulation? Is it harmonic order multiplication? Is it something else, like Hiraga distortion? Maybe. I don't know for sure, but I haven't given up.
To think that an IC like a 5532 or 5534 is all that anyone really needs, just isn't realistic.

Results folks, results! That is the key! Don't just wish away new ideas, because they do not easily fit into your vision of reality.

what about the 4558? That is a more realistic IC that used in typical audio equipment.
The AD797 is one of the best examples of IC design, today. Still, a discrete design can still have certain advantages over it, such as pure class A, full complementary differential folded cascode, all fet input. That should be more linear than any differential transistor input stage, and that is what I use, when I am able to.

I built what I previously described, as an open loop line preamp, the CTC. 350KHz open loop bandwidth anyone? Yes, I shamelessly copied Charles Hansen of Ayre, who did it first. I was chicken to do it, before I saw that Charles had gotten away with it. Why, because it is difficult to get GREAT specs with ANY open loop circuit. But what the heck! It REALLY WORKS!. I spec the distortion at .01%, IM or harmonic, at 3V balanced out. The distortion is, more or less, within this spec. With feedback, I could have gotten the same or better spec at 10V out, BUT it would not sound the same. How do I know? Because the JC-80 is a great example of a quality feedback line preamp design, using essentially the same parts.
Scott, you are far more sensitive than me regarding pre-'print-through' caused by records themselves. I have heard it before, myself, but I usually ignore it. Good to hear from you.

Feedback

For the record, both Charles [Hansen] and I prefer to build open loop circuits, if we can. We find them to sound better, all else being equal.
On my testbench is a preamp that is amazingly similar to what Charles Hansen builds (for good reason, since first I saw his design). It runs open loop. When I look at the harmonic series on a scope or FFT analyzer, it is amazingly 'pure'. This means that it contains very few higher order distortion products. Feedback circuits that I make with similar design concepts have more higher order products, not much perhaps, but much more easily measurable, than open loop designs.
Of course, feedback amps and preamps have their place, and are often necessary to meet a spec. This spec could be damping factor, THD, IM or some other spec. This does not necessarily improve the sound, and often tends to compromise it, if anything.
Now Charles might build an amp with a damping factor that is 'let's say' 30 or so. I might, with feedback, build a similar amp with an effective damping of 300 or so. Is this better? Probably no, but it looks good on a mid fi spec sheet.

Feedback

Now, when it comes to distortion byproducts, I don't know how to compare the two. I am pretty sure that a typical CFP has a lower output impedance than a dual emitter follower. This was the reason that Otala found that this was the best performing output stage for IIM distortion, which requires a low 'open loop output impedance' for lowest distortion. Personally, I am concerned with any kind of feedback, because it tends to multiply the intrinsic harmonic series, generating higher order harmonic and IM distortion products. I don't know if this happens with a follower, however.
I agree with your ' base stopper resistors'. Many people don't realize that dual followers can create so much phase shift at very high frequencies, that you can actually generate -R, or negative resistance. This means that oscillation is going to happen. + R in the input leads of the transistors usually cancels with any - R generated.

Hawksford error correction

I get confused about this as well, but I am pretty sure that it is NOT feed-forward. It is a local amplified feedback loop that might have some intrinsic advantages. I have never tried it, myself, but with a rock band power amp, it might be just the thing.
The reason for this is because, IF you have a fairly high crossover distortion (open loop) in the output stage, then this may be just the circuit to fix it, before applying overall feedback, if you wish.
My colleagues, Nelson Pass and Charles Hansen, apparently have tried it, and Charles, at least, doesn't like it much. This is a disappointment, but oh well, we just have to keep trying for better audio quality.

3.1 Harmonic distortion

Feedback and amplifier design

I started with first class op amp based designs, 30 years ago, to transconductance amp designs 20 years ago, and finally to 'pure' open loop designs, much like Nelson Pass and Charles Hansen now use. I have evolved, you should try it sometime.
By the way, you had trouble with complementary differential inputs? How, why? Because Doug Self didn't invent them?

It is difficult to make a power amplifier without global feedback, unless it is truly class A, which means low power and/or BIG heatsinks. Also, it won't meet THX specs, which is important in the home theatre marketplace.
The problem is that the amp needs to put out almost 28 times more voltage than the preamp. This means that it will make about 750 times more 3rd harmonic distortion (the lowest order distortion not balanced out), so the distortion gets very high, very quickly with increasing power
Now, I might be able to make a preamp with .01% distortion at 2V out. This would produce 400 watts from my power amp. However, my power amp, even class A, and built with exactly the same quality as my preamp, might have approximately 750 times

more distortion, so it would have 7.5%, at the very least, probably much more. Do these figures look somewhat familiar?
Still, listening from the first watt to 10W, I would suspect that the open loop design would be superior, as the distortion would be only .02-.2% or so.

1 volt into an amp (from the preamp) outputs to 28 Volts, so the preamp has an easier job. The square 28 is about 800 and third harmonic distortion will increase this much, all else being equal.
Halcro is an esteemed competitor. They use lots of negative feedback in everything that they design, not just power amps. 'Stereophile' puts both Halcro and the JC-1 power amp of my own design at the class A rating level. Both are global feedback designs, but Halcro takes it much farther than me. However: Martin Colloms in the June, 2004 issue of 'Hi-Fi News' criticizes the Halcro DM38 on several points, but we all agree that their distortion specs are lower than anything else around. Of course, you have to pay 3 times the price, and get 1/2 the power of my design, but the specs look great! So much for the 'cost savings' of a high feedback design.
The Halcro preamp did not fare much better in Martin Collom's review in 'Hi Fi News', April 2004.
My no global feedback preamp, the CTC Blowtorch, will never be reviewed by 'HFN' or 'Stereophile' because we don't have dealers, and our design cannot be considered for review, but it was reviewed in 'Ultimate Audio' which put it ahead of the pack. This is to be expected.

I will try again: It can be shown that second harmonic will increase directly with level, so: 10 times level eg 1V to 10V will give you 10 times more distortion.
Third harmonic will rise as the SQUARE of the level, so 10 times level gives you 100 times more distortion. "Thus for small distortion, HD3 increases 2dB for each dB in signal level" This is from class notes in a course in nonlinear signal analysis taught by Dr. RG Meyer at UC Berkeley about 30 years ago.
Found on p138 of 'Analog Integrated Circuits for Communication' Pederson/Mayaram, KAP
Also found on pp374-376 of 'Analysis and Design of ANALOG INTEGRATED CIRCUITS' by Gray/Meyer.
I hope that these references will make it more clear.
In any case, WHEN the preamp is at 1V, the power amp is putting out 28V. It is difficult, if not impossible to make a power amp without global negative feedback to have very low distortion (below .01%) at 28V. What about 56V? (the rated output of the JC-1)

Harmonic distortion in amps

The main thing to remember is that IF you can measure 3rd harmonic at some level, you can extrapolate the amount of 3rd harmonic as the square of the level at either a higher or lower level, just like you can with analog magnetic tape. 2nd harmonic is different, in that it changes linearly with level.
This is why, so far, I have avoided making a 'global feedback' free power amp.

Now, the REAL multiplier is 800. This is because of THX and its insistence that .1V in = 1W out. You can do the rest of the math.
Next, back to what I was originally saying: IF you have a driver stage running open loop with, let's say: 1V out = .01% 3'rd harmonic distortion, THEN the power amp, made with EXACTLY the same driver stage, BUT with an added output stage (presumed 0% distortion contribution) will be adding 800 times more distortion than the preamp to the final distortion. This is because of the difference in output voltage between the preamp and the power amp at any one instant. It isn't very practical to have such a difference between the preamp and the power amp contribution, especially for high power, home theater designed amplifiers. Maybe you can get away doing this with a 10W amp, but not a 400W one. That is why I use global negative feedback in my 50-400W power amp designs. Now, Charles Hansen and Nelson Pass have made amps that have little or no global feedback and their measured specs show what happens, no matter what they do, or how hot they run. Still, at normal listening levels, zero global feedback would be preferred, all else being equal.
To keep this thread 'legal' I might mention that this difference between second and third harmonic level changes was learned by me in class and in textbooks as well. With experience in using conventional test equipment, this is obvious, BECAUSE the gain settings on conventional test equipment change voltage in 10dB steps. SO, if I have an amp (or preamp) at one output setting, and I click the oscillator gain switch 1 position (10dB), then I see 10 times as much distortion, if it is primarily third harmonic in content and most of my designs are that way. It becames second nature, after awhile and that is why I kept repeating myself----because I couldn't determine what I had stated was at issue.
I doubt that you can get away from this formula in a class A system.

THD limit

It is a waste of time to talk about THD, UNLESS you also know the order of the harmonic(s) that are present. .01% 7th harmonic may be more annoying and even more audible than 0.5% of 2'nd or 3'rd harmonic. This has been known for about 70 years at least. It is just conveniently forgotten about by mid-fi manufacturers.

the facts are that it is easy to make an amp that measures very well at high levels, and poorly at low levels. Just make it class B, which usually generates some crossover distortion. High feedback will tend to hide distortion, but crossover distortion, if serious enough, implies a dead zone, that actually removes gain from the forward path at crossover. This makes things worse than normally expected.
Also, crossover distortion is often hidden in the residual noise (80KHz or so) of the test equipment measurement at low levels, and of course, is reduced in relative amplitude when measuring high signal levels, becaise the crossover distortion region remains the same level, and the ratio between the two is greatly increased, compared to low signal levels

In order to resolve low level distortion created by crossover distortion artifacts, we usually use some form of spectral analysis on the THD+N residual and reduce the noise floor, perhaps 20-40dB.
Generally, crossover distortion is separate from class B distortion, and rises proportionally at low levels. It is not monotonic with level.

First, push-pull class A design cancels even order harmonics, BUT local feedback, even created by another active device in series, will generate odd harmonics from the even harmonics; for example, in a differential pair.
However class B design, such as an output stage, DOES turn even order harmonics into odd order harmonics, because each output device only amplifies 1/2 the sine wave making up the total signal.
Personally, I usually chose to use push-pull class A and differential input stages for solid state. This reduces distortion in general, and makes for DC stable designs. The small amount of extra third generated is not very much, or very important.
However, tubes are another story. Usually, triode tubes can be made so linear that it is a toss-up whether they should be used differentially, except for special applications.
Third harmonic cancellation is another story. It is difficult to do. Still, pure third is not so bad. After all, analog magnetic tape had typically 1% third harmonic at operating level, increasing to over 15% on peaks, yet it could sound pretty darn good. Third harmonic distortion, and its IM products are still close to the music.
It is the 5th, 7th and 9th harmonics that should be removed or avoided at any cost.

Active devices usually naturally produce 'expanding' third, because their transconductance rises with output current.
However, even a small amount of local feedback, often necessary for temperature stability, etc. will convert a portion of the natural second harmonic into 'contracting' third.
If you are VERY lucky, you can find a true cancellation, but it usually isn't in a practical circuit.

3.2 TIM, phase, and modulation distortion

Slew rate requirements

Walt Jung and I have independly found that: .5V/us /Vpp is the minimum to really be safe with solid state amps. This is about 50V/us for a 100W amp.

base my recommendation for slew rate on what I personally measured 25 years ago with mis-tracking phono cartridges. They can really SPIT! I have seen measurable output at 500KHZ!
For the record, we have found that, in general, that a high speed power amp has many sonic advantages. It is best to keep them darn fast, if possible.

Just for clarity, EXACTLY 100W into 8 ohms would imply only 40V/us, but I have never designed an amp in the last 30 years with a slew rate of under 100V/us and I never will. It is just too compromising of the rest of the design.
Early 70's amps had slew rates of about 10-25V/us for the most part and they really sounded lousy. The first amp that I listened to in my own system, that really stood out, was the Otala power amp, which has 100V/us. I still have one today, in my office, and it still sounds great.
There are several approaches to understanding worst case slew rate. One way is actually measuring it. I have used a high speed memory oscilloscope to do this. Another way is to attempt to calculate it from available bandwidth, which is usually much more than 20KHz.
Many, not used this kind of audio measurement, might think that transients start at 0V and go either + or - , but actually the transient can start from the bottom of the waveform, go through 0, and end at the top of the waveform in 10us or so. This coupled with a 5 times multiplier to keep the slew rate mechanism from adding distortion, as it is not usually an abrupt limiting mechanism, then gives 40V/us absolute minimum and 50V/us as a reasonable minimum. As I said, I would still design at 100V/us or more, just because it is easy enough to do. I suspect that other factors such as open loop bandwidth, input linearity, etc, are equal contributors to audio improvement, but these mechanisms are what creates low slew rate in the first place, so reduction in dynamic phase distortion or PIM is a bonus.

How is it that a 100Hz square wave could possibly create slew rate limiting? How much ultrasonic energy is in such a wave. Yes, it can create SID or TIM. You don't need full voltage swing at ultrasonic frequencies, just the bandwidth to do so.

I have a square wave with a 1ns rise time. This is fast enough for me. However, for general testing, we often reduce the rise time to about 3-10us. This is because we like to be realistic about what actual sources can produce. Slower than this, is not realistic in all cases. Perhaps some of you will never see a risetime faster than 20us, and you can live with a lower slew rate. However, for reasonable worst case, 3-10us, especially with DVD and SACD sources makes more sense, along with moving coil phono cartridges and their effective bandwidth.

I referred to 0.5V/us/Vpp NOT 0.5V/us/V which can be interpreted to be only from 0 to max volt in either direction.

Why I have to defend 50V/us is laughable! You, I, Nelson, Charles, and dozens of other experienced designers have settled this years ago. Hundreds of measurements covering virtually all the existing amplifiers have been done, graphed and analyzed. For a number of reasons, a reasonably high slew rate is necessary to make a good solid state design, in general. It MIGHT be possible to make a pathological design that hard limits at a somewhat lower recommended slew-rate, but which operates virtually perfectly below its slew-rate limit. It might also be possible that you could perhaps make an automoble that was lightning fast in traffic, yet would not go faster than 80mph or so, but it would not be easy. Normally, the best handling cars go faster than 80mph, just because this also helps make their response effortless at lower speeds. It is very much the same thing with slew-rate and amplifiers.

I don't know if I can make better sense of everything that you are talking about, but slew rate is a LIMITING CONDITION that you don't want to approach. Do you need 500V/us? Couldn't hurt! Can you get away with 10V/us? Maybe with a bass amp, even a midrange amp, but not a full range amp, for best solid state audio fidelity.
Now, do some output devices limit slew rate? Yes
Do some input topologies limit slew rate? Yes
Today, with improved devices, the output stage should not limit the slew rate to any significant degree. However, 30 years ago, this was not the case.
However, the most significant limitation to slew rate is both the amplifier gain bandwidth, and the transconductance of the input stage. It can be shown that transistors have so much transconductance, that compensating for this lowers the slew rate below what is acceptable, in most cases.
When you try to make a fast amp, you usually also make it more sensitive to capacitance on the output. This can make an amp oscillate. Protecting an amp from output capacitance with an output coil can compromise the overall sonic quality of a power amp. This is why I choose to designing about 100V/us into my typical designs, today, even though I have designed faster amps in the past. I can remove the output coil, and still remain stable.
I hope that this helps.

Slew rate

Please keep the slew rate above 50V/us. Trust me.
A follower can be a problem with difficult loads, because it has phase shift at very high frequencies. Still, a follower is the best compromise for a power amp.

Slew rate

This is the situation, folks. There are many feedback related events that generate distortion. TIM (SID), PIM, load instability, and higher order conversion from lower order nonlinearities.
A sufficent slew rate, coupled with a linear input stage, eliminates just TIM.
Please remember, that 30 years ago, when we were wondering what was wrong with our audio designs, 0.5 V/us was considered by MANY COMPANIES as sufficent. This included tape recorders, studio boards, phono input stages, etc, etc. We had to FIGHT to get people to understand that simple slew rate was a worse case situation, AND that the distortion started building 5-10 times BELOW slew rate.
After 100's or even 1000's of measurements and giving papers at the AES, etc, people begrudgingly started to make faster IC's at low cost, as well as faster power amps, and the problem began to recede.
Then you get the 'academics' who want to take over, by criticizing previous work, and renaming the distortion mechanism.
And so it goes.

As far as I understand, two stage compensation is rarely used in most audio designs. Once, 25 years ago, I developed a discrete 600 ohm driver for Sound Technology that could do VERY LOW DISTORTION at 100KHz and below. I used 2 stage compensation in order to get a 100KHz open loop bandwidth in order to have MAXIMUM feedback available at 100KHz. So far as I know, this circuit resides in the next generation ST distortion analyzer, after my own equipment.

But, current feedback was used BEFORE voltage feedback, because transistors and tubes were expensive, and DC was not a problem with cap coupled designs. Big deal! Both ways will give you a fast amp, if you design it properly.

PIM

I do not understand why double pole compensation will eliminate PIM, except for the fact that it allows higher open loop bandwidth. The last time I used double pole compensation, I had an open loop bandwidth of 100KHz. Seems like a good idea to me, if this is the case.

It is difficult, at this time, for me to describe PIM, except that it is FM distortion of audio signals; rather than AM distortion, which is the normally measured distortion. I now have Barrie Gilbert's paper, but I must reread it carefully, before I can further comment on it. I will try to contact Walt Jung as well, since he has studied this paper and first recommended it to me.

In my practical experience, I use global negative feedback, most of the time, in order to make consistent, easy to make, products. In doing so, I always try to make a relatively simple, balanced circuit that is as linear and as fast as possible. I don't expect negative feedback to do much more than give me better techincal specs. than I need to meet THX requirements.

For me, this makes a 400W amplifier difficult to design, without global negative feedback. Smaller amplifiers, adjusted carefully, and using very high class A operation, can be acceptable, not using global feedback, so long as technical specs are not considered very important.

Let's just say that I believe that PIM or something similar is why many professional audio designers don't like to use negative feedback any more than necessary.

I spoke to Walt Jung about it today. We talked about local and loop feedback on input stages. Walt agreed that this would linearize the input stage and greatly reduce PIM. However, he also stated that that would still not make global negative feedback next to perfect, as far as he was concerned.

Walt and I, through experience, have found that op amps with high open loop bandwidth almost always sound better that op amps with low open loop bandwidth. PIM is a good candidate for why this is our experience.

However, it is possible that acceptable designs exist that use large amounts of global negative feedback with low open loop bandwidth. So far, some candidates for this are much more expensive than my power amp designs and comparatively priced with my best preamp designs.

Below a certain point, lower order distortion, 2nd, 3rd, 4th, don't matter to any degree, even though they may 'bias' an amp to sound a certain way, you know: 'musical', 'soft', or just something distinctive.
Higher order distortion, 5th, 7th, 9th, in my opinion are VERY problematic, although a little 5th may be unavoidable in solid state circuits.
Tubes have certain problems and advantages. I have found that if you can use their 'advantages' but exploit the characteristics of solid state devices, you can compete with tubes effectively.
This usually includes: High open loop bandwidth, low global feedback or even NO global feedback, and simple gain stages, and class A operation.
However, solid state can improve on tubes with 'direct coupling' between stages, complementary design for lower open loop distortion, lower input noise, and much higher peak output current.
I personally lived with a tube preamp for about 10 years, including a Dyna PAS 3X, Mac C22, and Marantz 7. I appreciate each of these preamps as being 'listenable' even today. I also worked at Audible Illusions in making their tube preamp. I think that the AI preamp is one of the best buys in the hi end audio industry, even now.
Now what do these preamps have in common to my later solid state designs?
Well, the Levinson JC-2 is really a discrete op amp phono stage, with a transconductance amp line stage. The line amp, especially, was designed after Otala paramaters: High open loop bandwidth and high slew rate, however it was also class A, and used a FET input stage, so that it could be direct coupled to the input pot. The JC-2 could use low value stage coupling caps, much like tube circuits, because of the input FETS. This turned out to be an advantage, as we found out later.
The Dennisen JC-80 uses all transconductance gain stages, but was able to remove global negative feedback on the phono input stage, direct couple all the stages, and create a balanced output.
The CTC preamp uses an open loop input stage, a low feedback second stage, and no global feedback line stage, all direct coupled, ultra low noise, and balanced output operation.
Now, over 30 years, what has been my direction? My direction in circuit design is to make direct coupled, class A, push pull complementary circuitry, that is low noise on the input and with low order distortion only at the output.
Does the CTC really measure better than the JC-2? NO! In some ways, the higher feedback levels of the JC-2 will give better measurements. Still, static measurements are not everything, and I know from experience that the CTC is the superior design.
What does this mean regarding this thread? Audio design is a progression of learning what works, and using it, rather that doggedly sticking to some pre-programed idea of what is important and what is not. Also, it is not an arbitrary assembly of different parts that go in and out of favor. Like tubes this years, solid state last year, and a hybrid design of both next year.
Audio is a vast marketplace, and is much like autos were, perhaps 100 years ago. There were many, many car manufacturers in 1904 or so, and each had a different concept of what an auto should be. The best ideas have evolved and the 'crazy' ideas dropped out of sight over time. It is the same in audio design, today. Many 'crazy' ideas will drop out of

sight over time, but this should not stop people from trying different ideas, we might learn something from them.

This is: The difference between local and loop feedback in audio stages. Yes, there is a difference.

Just because YOU can't find any justification for high open loop bandwidth, does not eliminate the fact that most real audio designs are limited by a real phase modulation due to a non-linear input stage. Look at REAL IC op amps, as they potentially have a serious problem in this regard.
It is true that a super linear input stage will reduce this problem significantly, but then I suspect that we would find another problem buried in the actual performance of these devices.
You equate subjectivity with experience, but it isn't necessarly so. Only a fool doesn't gather experience over time and trials, as to what works and what does not.

'Real' is very difficult, if not impossible for hi fi playback. All we can do is get the 'info' from the source to our ears.
If I listen to typical IC op amps, I hear a certain processed sound that appears to remove some of the 'info' from the listening experience. For example, I can't tell the difference between DVD and SACD reproduction on my Sony SACD-DVD player. I attribute this to the IC line amp, which I hope to change out with a discrete design one of these days.
Is it TIM? Is it PIM? Is it crossover distortion, or thermal feedback? I don't know for sure, just that IC's tend to remove subtle information from the audio source.
When it comes to 'Real' and EXCITING: I ALWAYS find that a quality phono playback will give this. Why? I don't know for sure.
I personally find CD playback almost always booring. Why, I don't know for sure, but SACD-DVD playback seems more interesting, but not as good as phono playback.
How many of you even listen to a quality phono system anymore?
Yes, I have questions. Professional questions that I must attempt to answer when I design new audio equipment. However, I use my listening experience and feedback from associates, to answer these questions.
Open loop bandwidth and the use of negative feedback is one of the most important questions that I must address. I still don't have any answer to this. I am now studying Barrie Gilbert's 1998 article that gives some real numbers associated with dynamic phase shift. It is complex and incomplete as far as details are concerned, after all it was only an overview of op amp problems, often overlooked by others. It looks promising to me that PIM generated by a TYPICAL input stage in IC op amps, causes problems with the listening experience.

I do know this: I have all discrete high speed electronics of my own design following the Sony player, up to the Wilson WATT speakers. Maybe I'm just deaf or crazy, but I can't easily tell the 'signature' of DVD vs SACD on my player. CD, I can immediately hear as inferior to either DVD or SACD. I still think it is the cheap IC in the line stage, that is the weak link in the chain. My associate and I have had the same problem before, when we added analog IC based products with our discrete products at CES shows.

Walt Jung is working to putting Barrie Gilbert's article: 'Are Op Amps Really Linear?' on Walt's Website. By working together, Walt and I are reconstructing the entire article for people to read for themselves.
Walt and I discussed its importance, just yesterday, and our conclusion is that negative feedback is problematic, and Barrie Gilbert has taken one of the potential problems into the public view. Matti Otala originally gave an even more complete 'qualitative' (just equations) paper years before, but Barrie has put forth a few 'quantitative' (with numerical calculations) examples into the scene.
The paper is now 8 pages of difficult reading for non-engineers. A challenging read even for engineers, as much has been glossed over for brevity.
I don't know what Mikek's problem is: Why ignore the obvious?

PIM

I would not attempt to equate the phase modulation with asymmetry, as it happens with symmetrical circuits as well, if the mechanism is simply non-linearity modulating the bandwidth with voltage level. If you have not read it yet, also look at Barrie Gilbert's article: 'The Multi-tanh Principle: A Tutorial Overview' IEEE Journal of SSC, Vol 33, Jan 1998. Walt may have it, but I also found it on CD rom, if you have trouble locating it.

PIM

It is important here to note that we have a problem of DYNAMIC phase shift that will be within the audio band. Now, can we use 'more linear' op amp input stages? Yes. Can we use 'less linear' op amp input stages? Yes.
Barrie's analysis presumes a perfect preamp, except for the input stage. The input stage is an ideal differential transistor pair. How typical of the 5532 op amp and many IC power amps, etc, etc. He sets the op amp for a gain of 10. Is this too much? Seems conservative to me.
IF you knew Barrie, like I know Barrie, he would not bother with this, unless it was seen by him as a problem. 32 yeas ago, when I tried to explain TIM to him, at a conference, you should have seen his face. I appeared to be pretty 'far out', to the point annoyance to him. Actually, today, he has a good understanding of TIM, but then it took a number of years before he realized what I was trying to point out, back in 1974.
Now, he is the 'champion' of PIM! Only 15 years after Matti Otala first wrote a 'qualitative' analysis of the problem (equations), that was not put into the 'AES Journal' because someone objected to it. And so it goes

If you have a very high open loop linearity, then feedback isn't really necessary. If you have poor open loop linearity, then you can have added effects, such as phase modulation, perhaps making a worse subjective result, than simply some lower order harmonic or IM distortion. My experience when looking at the open loop linearity of many IC op amps is that they are not very linear, i.e. their transfer function does not map as a straight line. This must create phase modulation, if the open loop gain is being modulated by the

nonlinearity of the op amp open loop transfer function. Feedback would be great, IF it were equal over the entire working bandwidth, but it almost always isn't.
Personally, I can't convey this concept as well as Barrie Gilbert can, so I give up trying to do so.

Personally, I have found that removing global feedback can make a better preamp, all else being close to equal My CTC preamp does not use global feedback, but it is difficult to make properly, as well.
The Parasound JC-2 preamp that is soon to be released, has lots of loop feedback, even though it uses exactly the same toplogy on the input stage. Parasound cannot afford to have any sort of 'marginal' specs, if it is to sell also in the mid-fi marketplace. I doubt that it will sound quite as good, as the CTC, because of previous experience with three other similar preamps that use almost the same topology.
My associate, Charles Hansen, does not use global feedback at all, even in his power amps. He doesn't do this for nothing. He knows that there is a difference in sound quality, when he uses global negative feedback.
I think the true test will be with direct subjective experiences between Halcro, Parasound, and Ayre. Each of these components are first rate, but there will be serious sonic differences to their ears.

Weighting of harmonics

How about a harmonic weighting factor of: A=(n-1)!/2
This will make higher order harmonics important very quickly. This was found in a '72-73 'Wireless World' article by Bob Stewart, now of Meridian. Works for me!

How about A[H(n)]=amplitude of the individual harmonics (n) The [!] is a factorial.

Trust me, it is difficult to get anything except a tube or class A FET amp to have an extremely low level of higher order harmonics, especially open loop and over extended frequency.
The reasons are:
Transistors are pretty darn nonlinear, and they have several different distortion producing components. These include: very non-linear Gm (voltage gain), non-linear BETA, and non-linear input capacitance (changes with voltage level on both the collector and base, referred to the emitter).
When you TRY to linearize them with local feedback (series resistor) you convert the even order harmonics into higher order odd harmonics.
If you try to use loop feedback, then you get TIM or FM modulation distortion, i.e. FIM from modulating the open loop bandwidth with amplitude changes with signal level. This is just with class A, Class AB or B is much worse.
It is a difficult problem. This is why we have developed sophisticated topologies in order to minimize the generation of distortion, over the decades.
In any case, the generation of higher order harmonics are not a good idea.

Most of our distortion is in the output, followed by the high voltage driver that has to develop the complete voltage swing for the amp. The input stage used to be a big problem when we used maximum Gm input stages. This was because the input stage would work harder and harder with increasing frequency, ultimately causing slew rate limiting, and even earlier, TIM (or SID). Walt Jung and Matti Otala have published reams of info in this, beginning in the 70's.
Today, we all degenerate our bipolar input stages, increase our gain-bandwidth of our amp, or both. Fets usually don't have to be degenerated in order to get a very high slew rate, because they are always lower Gm than non-degenerated bipolars, and they are more linear as well.

3 Audio

Amplifier design

Actually, each amplifier design has a number of tradeoffs. It is fairly easy to make an amplifier that has almost no distortion. This is done by negative feedback, both local and loop. It can also be done with feed-forward, or the Quad 'current dumping' type circuit. Each designer must decide what sounds best: ultra-low distortion, single ended, open loop operation, class A, etc. No one approach is necessarily better in every way, from the others. This is about all that I have to say on the subject.

Symmetry

complementary symmetry is NOT perfect, because the comp. input devices are not perfectly matched, BUT look at the alternative. Single differential designs are OK on their input, BUT what about the drive for the second stage? Comp. Differential gives you almost perfect push-pull drive for the output stage. This criticism of the lack of perfect mirror image is lost in the advantage of one drive device turning on, while its complement is turning off from the opposite rail. This is invariably better as far as open loop distortion is concerned.

in many cases, PNP transistors and N channel fets have an advantage. Why, because of the difference in mobility between holes and electrons. PNP has N in the base (sensitive region), and N channel fets use it for the channel. This is why, I am told. However, SOME NPN transistors have very high betas, better than pnps for the most part, and there must be some reason for this. The usual difference that can be easily measured between devices is the rbb' or intrinsic base resistivity. PNP's usually have about 1/2 the resistance compared to NPN's.

Balanced connection

Balanced business is mostly a myth. My Vendetta phono input is one of the quietest preamps in the world, and it is single ended input. The only thing that I have to worry

about is putting a power supply next to the preamp or the wires that come from the phono cartridge.
Of course, balanced may be necessary in some applications, such as running a low level signal from a soundstage 50 meters into another room, or a remote mixing board. Also, if you have light dimmers installed and running, balanced might help. However, balanced is overrated. This I know from personal experience.
Dave Wilson is using the pre-preamp version (SCP-1) in his home reference siystem. You know, $200,000 in cost, with BIG subwoofers. If it hummed, he would certainly tell me so, and I just spoke to him yesterday.
Hum is usually caused by no external shielding, or ground loops within the chassis.

Crossovers

You can actually have essentially zero delay with a 6/dB xover. Trust me, or else, read up further

Loudspeaker load

First, a loudspeaker is NOT just an inductor. In fact, that is a very small part of the speaker circuit. The moving loudspeaker is a resonant SYSTEM all of its own that CAN be represented by an 'equivalent' R, L, and C model. Also, loudspeakers are microphones and pretty good ones at that, so any sound in the room can be put back into the loudspeaker an emf generated across the speaker terminals. What about a resonant cabinet? What about a port?
Loudspeakers may be 'simply' modeled as an equivalent RLC circuit, but that is not their complete response in the back EMF.

Loudspeaker distortion

32 years ago or more, my associates and I wondered why speakers could have so much distortion and amps could not! Even more, some amps with LOW distortion sounded worse than amps with high distortion. It has to do with extra distortions generated by the amps, BUT not measured with normal audio measurements. Feedback IS an important factor.

Microphones

Most audio designers are practical people. We have addressed the distortion in loudspeakers and microphones. We know, from experience, that preamps and power amps STILL add something, if not carefully designed.

Microphones are usually really low distortion at realistic input levels. Of course at 120-140 SPL they can measure some distortion, depending on their quality and sensitivity. This can be computed backward to estimate distortion at normal input levels, 90dB or so, by considering the elements class A in operation (no xover distortion) and noting the dominant harmonic (usually 2nd).

Transformers

I have worked all my professional career to REMOVE transformers and caps from the audio circuit, but most examples in the marketplace still sound worse than my old Telefunken radio. For example, I have a Class D amp of similar power and when I use it in the same application, it is NASTY! Virtually unlistenable. I was even a consultant, after the fact, on the design. There wasn't much that I could recommend to improve it, however.
Folks, we have to keep our priorities straight. We should try to make good sounding designs, not just good measuring ones, or just theoretically possible ones. You know, infinite feedback means 0 distortion

Loudspeaker current drive

For the record, current drive is not unusual for motor drive applications, and has been used for many decades. However, loudspeakers, under some circumstances, could benefit from current drive, BUT NOT typical speaker systems.
Also, I could care less whether SE is on this website or not. I like this website, because it is fast moving, and many inputs post interesting schematics and other references. I would prefer to keep on subject, if possible, but I'm sure I also have diverted from the original subject on many occasions, over the years. We were having an interesting discourse on differential input stages, but now this is pretty much lost in the noise of other inputs.

Null test

the idea of the NULL TEST is not a new idea. Walt Jung and I used it comparing capacitors with a AD534 instrumentation IC that has very high common mode rejection. What is wrong with this test? Well, it doesn't separate LINEAR from NON-LINEAR distortion. What does this mean? Well, ANY time delay, phase shift, dielectric absorption, etc will OVERWHELM the test itself.
Yes, you can attempt to compensate for some of the linear distortion, BUT not all.
Is this possible with digital subtraction? Yes, John Meyer and I first used it 30 years ago to measure loudspeakers. What is the problem with this test? Well, you need a lot of bits of resolution in order to make it useful. Remember, you are looking at a DIFFERENCE, and the residual must be error free to be useful. Perhaps a 24 bit system, with really good supporting electronics, MIGHT make it possible to measure audio differences, but I have not yet tried it. Perhaps someone else has.
Critics and cynics, get out of your armchairs! Get to work making the perfect single measurement system. Until then, please live with the fact that no ONE measurement shows EVERY deviation from ideal, in audio electronics.
With all due respect to David Hafler. His amp may have passed his test, but that amp is not considered 'perfect' by any standard, and is usually 'upgraded' by amateurs like yourself, with noted 'improvement'.

Digital staff

Class D! Give me a break! I get class A reviews, BECAUSE I care what the actual sound quality is. No more, no less. Up to date? Clue me in. When I really want to hear quality, I put on a record, sometimes one found in the trash of someone else. Digital essentially sucks!

When DVD or SACD can do 'Live Dead' then I will convert. It is more 'pure' that the other recordings that you mentioned. By the way, SOMETIMES digital sounds darn good, even great, but not often, to my ears.

Digital mostly sucks, I think that the 'poor digital transfer' is just an excuse. How hard can it be? Negative feedback should be reduced or avoided if practical. Preamps sound better without global negative feedback, at least in my experience.

3.1 Input stage and preamps

Double vs single diferential

Complementary differential or Double differential was first used by Jon Iverson of Electroresearch; and me, while I was employed by Ampex in 1968, independently of each other. Later, Southwest Technical Products published and advertised the first commercial amps using this concept. GAS came out a few years later.

I have used jfets rather than bipolar transistors for the last 30 years. All the Levinson stuff was jfet input, usually complementary differential jfet.

2SK389 / 2SJ109 complementary monolithic low noise JFETs
2SK170/ 2SJ79 complementary high gm low noise JFETs
2SK364/2SK104 complementary very high gm JFETs
2SC3381/2SA1349 complemantary monolithic low noise BJTs

The j109/389 are still available. We use 100's of them every day. The Japanese manufacturers run out of and 'discontinue' a part, then they get so many orders that they have to make another run. Just keep your eyes open on the internet and do a periodic search for the part on Google, or another resource. Levinson (the company) went away from fets in most designs, mostly for practical reasons.

We use 389/109 in all of our Parasound power amps. They are made in Taiwan, and I don't know the resource for them, but I buy them fairly often, BUT I don't go to Toshiba directly.

RIAA preamp input stage

Normally use the highest current to get the lowest noise. If you HAVE to starve the fets, then the GR parts may work best, but the BL parts are actually better operating at nominally higher currents. Run at high current, if possible.

Phono preamp

First, most MM cartridges today don't have as much inductance as early types. Think it through. As the effective stylus mass is reduced by improved design, then very high inductance will tend to roll-off the cartridge too much. This, of course, must be balanced by higher effective output, per turn, to make up for any lost inductance. Most MM cartridge manufacturers have addressed this issue, so that it is not much of a problem, anymore.
Second, NEVER put a 100 ohm resistor in the gate of a 2SK147. You will compromise its noise significantly, and increase high frequency non-linear distortion.
Third, it is usually best to use a FET input op amp or discrete design, so that no added current noise or DC base current is put through the phono cartridge.

MC cartridge load

The summing resistor to lower impedance was part of my JC-1 patent, first submitted in 1974. We have gone away from short circuiting the input for MC's, because it usually sounds worse than a higher impedance in modern MC cartridges.

his is an early idea, that we ultimately 'scrapped' over the years. I contacted EMT and Ortofon about optimized cartridge loading about 30 years ago. Their responses were vague. Also, AJ Vandenhul recommended low Z loading on EMT cartridges that he retipped.
As the years went by, optimum loading has tended to go to higher values, rather than lower values. Why, I don't know for sure.

Noise in BJT

Noise has several sources. One is base resistivity or Rbb'. This can be 2-400 ohms. The very best NPN transistors have about 4 ohms.
Then the Voltage noise generator. For transistors, it is .5/Gm = 13 ohms at 1ma, 1.3 ohms at 10ma. For fets, it is .7/Gm or.7/.035 = 20 ohms at 10ma.
There is also a current noise generator, which will be neglected here.
Now what if you ran the transistors at 10 ma? Then the transistor would be best, BUT it would be made more noisy than necessary by 1/f noise and added current noise.

The Levinson JC-1AC was developed 29 years ago as an AC version of the JC-1. However, I made a 'fatal' mistake. I used a LOW Z input by using a grounded base connection as the input. This overdamped the MC cartridges. I proved this by rewiring the design to be switchable to either 100 ohms input or grounded base (very low Z) and listened to the difference. 100 ohms input won the contest.

For the record, we were going to use a design similar to this in 1974 for a microphone preamp stage. It was first published in one of the electronic engineering mags, either 'EDN' or 'Electronics Design' back in 1974 or so. I think that it is a very elegant circuit.

The JC-1 AC sounded 'overdamped' with my phono cartridges. I am not criticizing the rest of the design that much, even if I don't use it anymore.
Now, this thread has all sorts of people 'discovering' summing type inputs and while this is OK, it can also be a problem. For example, about 20 years ago, I designed my Vendetta Research SCP-1 (SCP-2 as well) input stage with a link that could be changed to have a summing type input. In future designs I have thought to eliminate this link because it didn't seem necessary for modern moving cartridges. However, JCarr and others have found low Z to be a good load for moving coil cartridges in some models. This is interesting to me, but I do wish that I had a specific model MC, or a design parameter to follow in order to determine what direction to go.

I don't know what the exact mechansim is, but I was once told that the magnetic assembly is important. Hi Z windings certainly would 'tend' to increase the optimum loading, all else being equal, but many MC's are still low Z and apparently, in one extreme case, best loaded with low Z. Go figure! Where are the phono cartridge designers, now that we need them? For some reason, intrinsic cartridge resistance seems to dominate the frequency response over the audio bandwidth. We don't actually know why MC cartridge loading actually makes a difference, but it does. Only cartridge designers could help us here.

Input Stages

Except for certain situations, it is better to use fets rather than bipolar transistors for low noise input stages, including MC inputs. This is because fets have almost no bias current and do not increase in noise with higher impedances. Paralleled transistors can be VERY QUIET at nearly short circuit impedances, but above about 100 ohms or so, can become more noisy than a single transistor. This is because the noise current generator (shot noise) becomes more important.

Paralleled 2sk146 fet pairs are 'overkill' and will actually add high frequency distortion (Nonlinear gate capacitance). Better to have a single 2sk389 pair.

Noise in BJTs and FETs

A low noise bipolar transistor has 3 significant noise contributions. One of these r(bb'), which is the intrinsic base resistance, can vary from 2-400 ohms, depending on the device. Of course, the MAT02, which is composed of 10 or more paralleled devices, has a low r(bb'), but interestingly enough, not the lowest in the industry. This would go to a PNP device from Rohm, Hitachi, or even Fairchild.
The second noise contribution, that is collector current dependent, is the second stage shot noise. It ALWAYS measures at the noise equivalent of 1/2 r(e). r(e) is always 1/gm which is: 26 ohms at one ma. 260 ohms at 100ua, 2.6 ohms at 10 ma, etc.

Therefore at 1 ma: [second stage shot noise] is 13ohms equivalent noise.
The third component is base shot noise and is more complicated to calculate, because it is sensitive to source impedance and any resistance in the emitter leg as well. It is also sensitive to the base current, so it is inversely proportional to the BETA of the device. The MAT02 has a beta of 250 or so, which is respectable, but still low enough to insure some current noise in a practical circuit, although still fairly low with a source below 100 ohms or so.
For the moment, we can ignore the input shot noise.
Well the best that can be done at 1 ma is: 13 +r(bb')[which at best is 2 ohms] so we could, in theory get 15 ohms equivalent noise or .3nV/rt Hz. Pretty good, but the MAT02 shows 1 nV/rt Hz, so it must have an r(bb') of approximately 45 ohms in order to make the results match the theory. Perhaps it is somewhat better than that, I know that some matched bipolar pairs are, but they degraded the performance on the spec sheet.
However, a 2SK389 monolithic fet pair can easily match this performance, but you would have to run it at 10 ma or so to get best results. A 2SK146 matched pair can easily beat this by another 3dB.
The important thing about FET's is that they have almost no input stage shot noise, so they do not get noisy with input impedances over 100 ohms or so, like bipolars do, when running at 1 ma. This is a great advantage in most cases. Now folks, what happens if we run the bipolar transistor at 100ua, instead of 1 ma?

Don't confuse things yet with the actual second stage. Let's look first at the first stage. Once again, the first stage device has several noise sources inside it. What I call 'second stage' shot noise is from the collector to the emitter, rather than from the input current (first stage). You could also call it output stage shot noise, does this make it any clearer? This has only to do with ONE device, not other devices.
Ultimately, the tradeoff is between the input current related shot noise and the output stage related shot noise. Reduce one, increase the other. These is where the concept of NOISE FIGURE which is defined by dividing one noise source by the other to find the optimum source resistance is derived.
Another factor is the tradeoff between very high beta, which would REDUCE input stage shot noise by reducing input current, and r(bb') which is the intrinsic base resistance, as it tends to increase with very high beta devices.
What a hassle! There are some fairly ideal devices out there, but most of you could not purchase them easily. I don't know why they are not more available.
With an inductive source, IT IS IMPOSSIBLE to have an optimum noise figure with a transistor over a range of frequencies, but with a transformer and a resistive source, or just a higher value resistor source alone, it is possible to get an optimum noise figure with a bipolar transistor. Noise figure with a fet does not really apply at audio frequencies, in general, because the noise current is so small, that the optimum matching is in the megohm region, but under these conditions a good fet will beat ANY transistor.

As idle current is dropped, what we define as 'voltage noise' will increase by the square root of the current change. The input current generated 'shot noise' contribution will reduce by the square root, IF the BETA remains constant. However, the beta usually drops, so the current noise will NOT drop as fast, as the voltage noise increases. Enough for now.

I design around 2 ohm sources, worst case, so I need everything that I can get. But then, I don't need to buy step-up transformers, or use cheap moving magnet cartridges. I went from bipolar to fets, in 1968, 35 years ago, when I made a quieter reproduce stage with fets than bipolar devices operating between 50-100ua, even then, when I worked for the audio division at AMPEX.

It has been my interest to bring some of you up to date on how noise is generated in solid state components. I deliberately left out additional stages and 1/f, as well as popcorn noise, because they are relatively low in good quality parts at audio frequencies. RF, or test instrumentation, could be another matter entirely. Many of the contributions and references have been as good as I have seen anywhere. It is pointless to make a noisy circuit, if it can be avoided with a little knowledge of what is important. One thing that I often see is a relatively large resistor, (over 100 ohms) in series with many preamps. This can be good for RF control, but it can add significant noise to a phono stage, unnecessarily.

Input stage design

The 100 ohm resistor is in series with the input path and generates 100 ohms of noise which is a little over 1nV/rt Hz. Why is it sometimes there? It can be part of a feedback return path, an RF 'stopper' resistor, or used to match the 2 differential inputs with the input resistance in order to get better tracking of the two differential transistors by balancing the base current drops at each input.
Is 100 ohms a potential noise problem? Sometimes. For example, in one of my phono preamps, it would add 10dB of added noise, quite a lot. For a cheap IC op amp input, it might be barely measurable. It just depends.

MC preamp design

About 1/3 of a century ago, we used a pair of 4401-3's to make a moving coil circuit. In my patented approach, we did NOT need either input or output caps. However there were other problems, especially with higher output mc cartridges and I moved on to fets. Low noise Toshiba fets are at least as quiet as a 4401-3 pair, BUT they requre several Ma of current to work successfully. They also usually require more voltage as well.
Back in 1973, Mark Levinson and I paralleled 4 pairs of 4401-3 parts to make the input low noise enough for an Ortofon MC cartridge like yours. We also ran about 8 ma per channel, so we had to use D cell batteries, and even they ran down, all too soon. This AA battery stuff is a big compromise, because if you STARVE the transistors, the battery will last, but the noise will go up big time! Working fets in their 'triode' region will not get you the gain that you need. I wouldn't worry too much about the leakage of a cap, just use a good one. Try to use Japanese or German caps, if you can. Cap size matters! IF you use too small a value (like 10uf) the transistors will get VERY noisy below 1kHz. Not good. I would recommend a simple 2sk170 Toshiba fet running at a few ma and even up to Idss

of the part. Common source or common gate will both work. I would probably use a 10 ohm resistor with a 2sk170 fet and direct couple to the gate. I would NOT bypass the 10 ohm resistor. I would set the load resistor, wherever I get the most voltage swing. I would probably start with 6V, but 12V would be better. Maybe a rechargable gel-cell battery pack. Either a 170 bl or gr will work.

Here is a list of conditions that are both impractical and problematic.
1. Balanced input may be a good solution, BUT you always make a noisier input stage with balanced, because 2 devices must be in series to make the input stage. You always lose 6 dB, compared to using the same input devices in parallel.
2. Batteries are good, because they are clean (mostly) and they prevent ground pick-up, due to the fact that they are floating.
3. If you do make a balanced input, it will hum, unless a balanced lead from the cartridge is used with an XLR or equivalent plug is used. Single RCA's will not work.

Will a small resistor 10-50 ohms raise the noise floor?

Of course it will significantly increase your noise. Why bother to use large area fets and put them in parallel, if you are going to use an input resistor with the same noise that you just removed? In this case, you have to work without an input resistor, at least anything more than 10 ohms.
My circuits run open loop, and they have the equivalent noise of a 10 ohm resistor. Your results are closer to a 50 ohm resistor, maybe much worse if you use any resistor at the input or as local feedback.
Look at the construction of a 2SK146. Does this make sense to you? You will find that two 2SK147's are put together face to face. Guess what? One has to be upside down. Still they work, still they track. Why worry about it? The same goes for the 2SK170, and many other devices.

JFET noise

Never deliberately add gate resistance to a jfet. It just makes it more noisy.

Noise Figure and transformer coupling

Noise Figure is almost useless with audio jfets. I know, I know, they still put it on the spec sheet, BUT it is a hangover from low noise bipolar transistors, where it really makes a difference. All of the fets that you mentioned are pretty quiet. This means that they have low voltage noise. If you are transformer coupled, then adding gain to the transformer will give you a better noise figure, because you are then operating at a higher input level. The tradeoff is the output impedance of the transformer, which means virtually NOTHING to both tubes and fets, BUT is VERY important with bipolar transistors. This is because bipolar transistors have an extra noise source in the base emitter junction that will go bonkers when you have too much input source impedance.

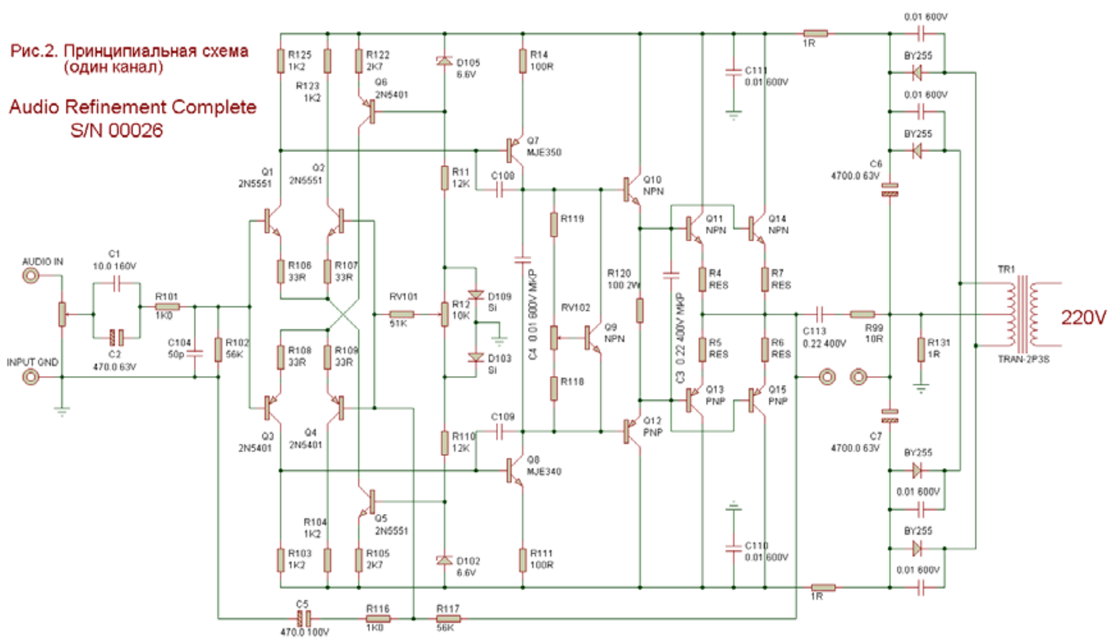
For the last 30 years, my total input noise has been equivalent to a 10 ohm resistor. Try that, with added resistance on the input.

Noise contribution of the first stage

First, we should understand what makes a noise contribution to an amplifier stage.
In a transistor, it is the second stage shot noise which is effectively r(e)/2+ Rbb' (which is a base resistivity common to all transistor bases). Rbb' can range from 2 ohms to 400 ohms. Selection of a high Rbb' transistor will spoil things immediately. The only real reason to parallel input devices is to lower the effective Rbb' with a 'transistor array'.
The second important contribution is the input resistor, put there for RFI, as a rolloff, or just for fun. This will completely compromise the input noise, if you are not careful.
The third contribution will be the effective NOISE GAIN of the current sources used as a load.
Another contribution can be from the differential pair current source, which will add its noise to the input, UNLESS the second stage has common mode rejection.
These are points to ponder. I have not done a full analysis of the circuit in question on this thread, but it would be worth a computer simulation to see each and every of these effects. Good designing

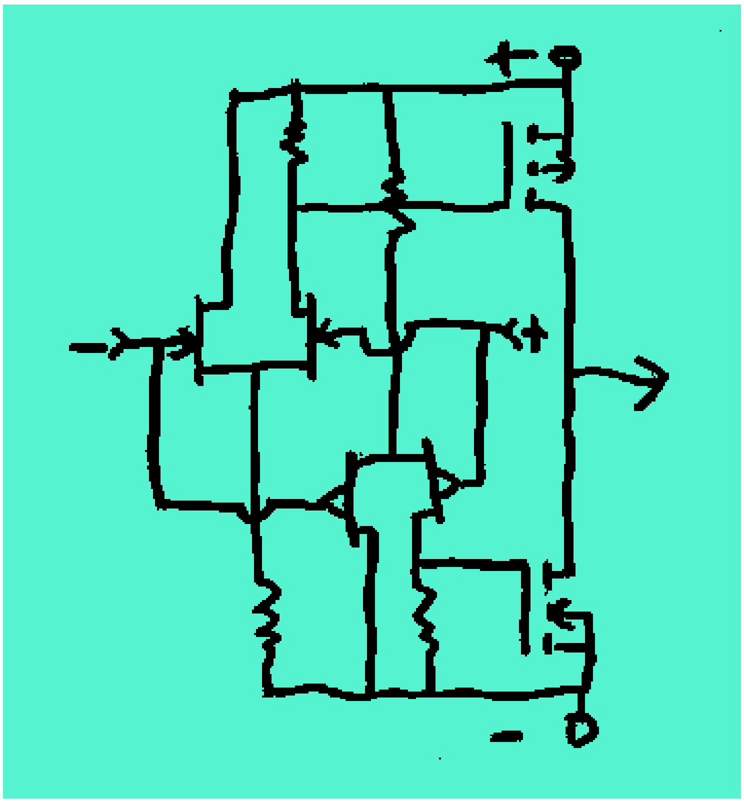
For the record, I usually design at 0.4nV/rt Hz, 20 dB below what is considered OK here. An example of one of my older designs, the Vendetta Research phono preamp, which I stopped production on about 12 years ago, is designed to 0.4nV/rt Hz and is apparently in 'Hi Fi Plus' in a recent issue. Even the Levinson JC-1 had the same essential noise level as the Vendetta Design, and first came out 30 years ago.

Single or dual differential



The complementary differential has lower distortion, all else being equal. If you build a design like the comp symmetry example shown, just put a large cap across the second stage base to the nearest supply. Either side will do. I found an increase of distortion of 5 times in an example that I measured more than 30 years ago.
Think it through. First you lose GAIN (6db) because you have only one working input stage. Second, you INCREASE even harmonic production, because you are not equally driving the second stage transistors. Try it and see. Don't worry about the intrinsic mismatch in N vs P transistors or fets, it still is better to use both together.

I would like to make the case for complementary differential topology:
It isn't the INPUT stage that is lowered in distortion, it is the 2'nd stage which is usually single ended and has to develop almost all the gain for the amp, which improves. This problem was first addressed with 'bootstrapping' using a cap connected to the output of the amp to give positive feedback and increase the driver load impedance. The next approach was to use a constant current source as a load, favored even today by Doug Self. Finally, the equal driving of both driver transistors, either with a current mirror, or with a complementary differential input.
I have used each of these approaches over the last 35 years, and personally, I prefer the complementary differential fet input. Don't tell Doug Self, but fets actually work darn well as input stages, and have many advantages, such as no need for an input capacitor, and very high slew rate operation, without any noise tradeoff. Also, they tend to be more RFI resistant, because their input diode is off, rather than conducting.
While I have the greatest respect for what Doug Self has published, please don't box yourself in a corner by thinking that that his input is the only or necessarily the best approach to circuit design.

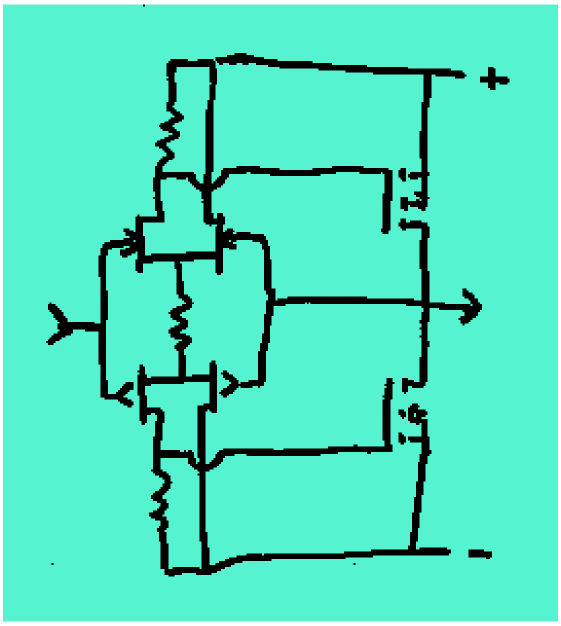


You are close, hitsware. This design concept is about 30 years old, as it was first used in the JC-2. Now, how can we improve it?

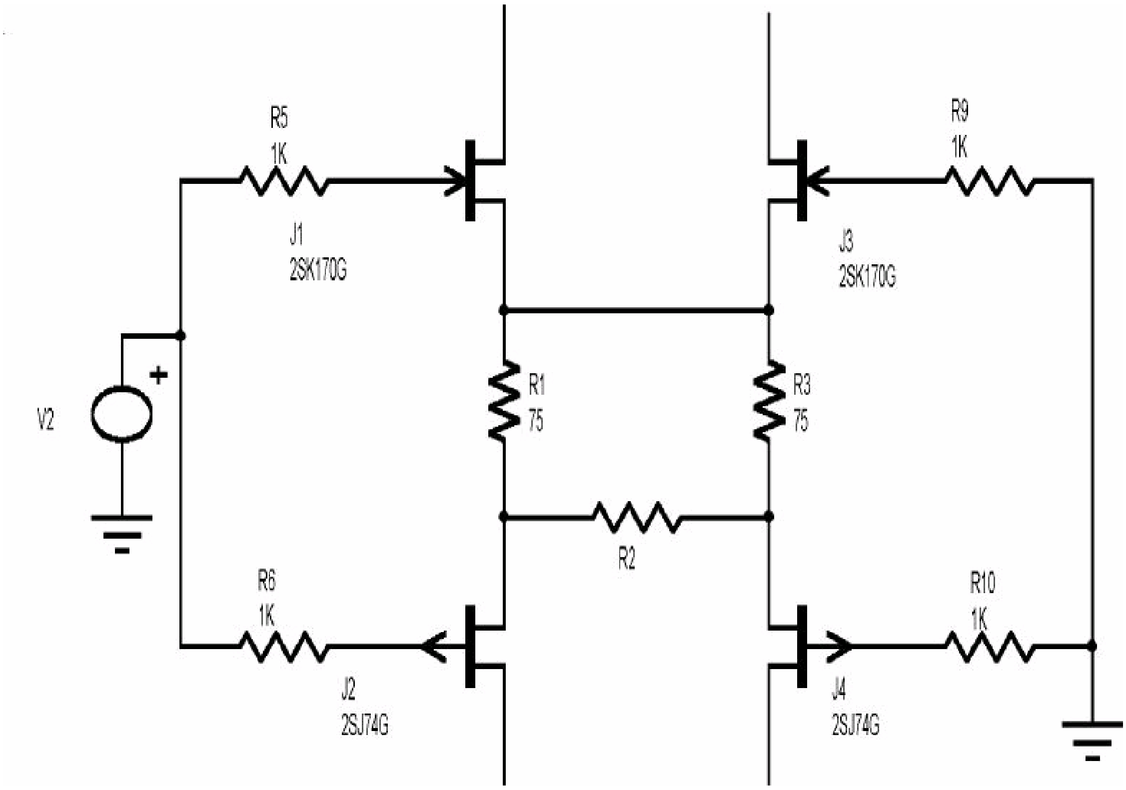
First, think about removing ALL the current sources and just using 1 resistor between the source pairs on the input. Second, this is actually a very stable design, because it has almost no gain, but 1/2A Hitachi devices can be problematic, because they are not well matched in this situation. The P channel looks like a triode, but the N channel looks like a pentode. Too much 2'nd harmonic. How do we fix this?

>Cascode the output pair?

Yep, makes a world of difference, in this case. This is essentially the 2'nd stage for my Vendetta Research phono preamp.



This makes a pretty lousy power amp, BECAUSE there is no loop gain with a low Z load. Makes a darn good line amp, however.
I even made the Grateful Dead line driver to drive the final mix from the mixing board to the stage with a similar configuration. All else being equal, fets are better than bipolars, but sometimes bipolars are useful in this circuit.



biasing has problems. Not in concept, but in execution. The resistor values are too large, and the differential gain is asymmetrical

Gate resistors are most probably optional on this design. Why? I don't know why you like gate resistors of jfets. All it does, in this case, is to make the input more noisy.

Personally I don't like to use much series resistance in the base of Jfets, BECAUSE it increases the noise of the entire unit. SE is correct, if you want to make an input filter, then put 50-200pf mica or film cap to ground to have a LINEAR capacitance at the jfet input. Then you can reduce the input R to 1K or so.

I felt that it was not a 'universal' schematic, because it lacked placement of a source degeneration resistor in the N channel rail. This is one specific case with using J109's and K389's - the difference between Gm's before, in these two devices. However, if you were using K170's and J75's, the difference would be less, and almost unmeasurable. Personally, in a simple circuit like this, I prefer to use high Idss types and run them as hard as possible. This includes even slightly overbiasing the input devices in the forward direction, under transient peaks.

I have NOT worked with Mark Levinson since 1976. Therefore I did not design the 6,7,etc products produced later. However, most design concepts are based on 3 or 4 design topologies. The JC-2 used 2 of these topologies. Some of these later ML designs use either these 2 topologies or something similar in many products.
I am neither for or against negative feedback. I just don't like to use an output stage feedback pair inside of a global feedback loop. There is justification for this that was debated extensively on a previous thread, somewhere.

The MAIN advantage of comp-diff input is that the SECOND STAGE is symmetrically driven. This reduces the effort of just one device having to drive the output stage, and effectively doubles the available class A current. It also gives a doubling in the open loop gain, without decreasing the phase margin, and third, with bipolar transistors, it can cancel the input bias, so that an input coupling cap is unnecessary. This offsets the extra parts count.
There are other advantages, especially with FET input designs, but this should be enough for this discussion.

Many advanced audio designers use techniques similar to or better than the concepts described in IC designs in previous threads. Often, we invented them first, such as the complementary differential input stage.

I doubt that Erno and I have too many differences of opinion. We have known each other since 1975. We have compared notes.

Complementary differential

Bob Stewart of Meridian is a colleague. He and I first met in 1972 at a Londen AES session with Baxandall giving a talk. He invited me to a party at his house, and we listened to his system. First, we had discussed the problem of E-I output limiters and sound quality. While we were listening to music, Bob took some snips and cut out the protection. The sound improved significantly. Then, at the end of the night, I gave Bob the secret of the complementary differential input stage. After all, I had already used it for 4 years, and Southwest Technical had discussed it in print in 'The Audio Amateur'. Well, in 1973, Bob wrote a great article for WW, but at the end of the article, it was mentioned that a new audio circuit was under patent review, and that it was a significant improvement. Guess what the circuit was? I made a telephone call to Bob and accused him of trying to patent the circuit that I had given to him. He admitted to it, BUT said" The patent didn't get through, so we can still be friends. " Oh well! If you are at the AES, SF in Oct, maybe we can all go to dinner and discuss this and other things.

Bob Stuart is a sharp guy, but the really interesting folks were Michael Gerzon and Peter Craven. (I think I got their names right). They developed the lossless data compacting scheme. Maybe Peter will be coming over to this AES, haven't seen him in years. Trust me, he is just as crazy as I am.

Power amp input stage design

First, what point is there to go to extremes to reduce the noise of the power amp? It is the last chain in the link and the vast bulk of the noise in your audio system will come from the preamp and earlier sources.

Second, using a very large, expensive, fet on the input is great for MC inputs, but the inherently high non-linear capacitance in the large area input fets will give you more high frequency harmonic distortion than a more optimum geormetry like the 2SK389. Noise difference? Almost unmeasurable in a power amp design.

Cascoding would reduce the distortion, but not remove it completely.

Paralleling FET's is what I do for MC inputs, but it only makes the situation worse for distortion with a power amp, or even a line preamp.

Think it through, look it up on the internet, or give thought of what audio designers have already stated on this thread and others.

Gate stoppers

I agree that a design with a mosfet like this should have a discrete stopper resistor directly connected to the gate, and that the gate should be clamped with a zener diode to protect it. I once had a similar problem with a j-fet follower, interestingly enough. In this case a fellow designer returned an electronic crossover that I had designed and built for repair. I found the problem and then added protection zeners to the gates of the input part. It came back a second time, and I had to put my foot down that there was something wrong with the high fi that it was being used in. Eventually they found the problem. In a case like this, a mosfet would have blown immediately.

I have not used (as a rule) gate stopper resistors with jfets, but I have had lots of parasitic oscillation, on occasion, with jfet followers.

The most troubling case was with my Vendetta Research phono power supply. Dick Marsh had been asking me to use his new, improved caps. I was already using tin foil (RT) polystyrene caps from Rel, so I did not see the point. Well, a customer of mine, (and a good friend) had Michael Percy (sp?)modify one of my preamps with Dick's caps. He only had 200V units available and my layout was for 100V. This meant that the caps 'hung over' the board. Guess what? The jfet followers oscillated! They oscillated at about 100MHz. M. Percy did not have a fast scope, but I could see it easily. The fix was to put a 100pf mica cap in parallel with the Marsh caps. It worked well, but perhaps if I had gate stopper resistors installed, I may not have seen this happen.

Transformer input

I don't wish to be a bore, but I just read some previous comments on this thread. First, we have to separate the peak voltage possible from a step-up transformer at the input of a mosfet, with the possible resonant Q of the transformer; and a normal input, even static. Static breakdown is very sensitive to the capacitance it must charge. A normal input will seldom exceed 15V or so. A transfomer could potentially 'ring' and punch through a gate. Just thinking worst case.

Current mirrors

You folks have the 'current mirror' concept right, BUT they can be a problem by amplifying their own noise. Degeneration resistors in the devices closest to the supply would help keep the potential noise contribution lower.

I prefer resistors to current mirrors, especially on my input stages. The high impedance of the current mirrors lowers the open loop bandwidth and actually puts stress at high frequencies on the second stage as far as nonlinear distortion generation is concerned. WHEN, I HAVE to worry about significant distortion reduction, I have found that complex current mirrors work better than simple current mirrors to reduce distortion over a broad range of frequencies. Sometimes, I will use a current mirror in a second stage for practical reasons.

I forgot to add that there is one other SERIOUS PROBLEM with current mirrors. They amplify their own self noise! This is very important and why I always use emitter degeneration resistors, unless working at extremely low power supply voltages. This greatly reduces the active gain of the mirrors and the extra noise can be minimized. Resistors have noise too, but no active gain to make it even worse. It took years for me to learn this, so don't be too surprised if you haven't thought it through, yourself.

Added noise is because your second stage current mirror SUCKS noise-wise ;-). Use larger value current mirror degeneration resistors in your design.

3.2 Output Stage

Mixing output power transistors

This is dangerous. However, with a one-off, it might work OK. Right now I am struggling with current hogging with the SAME type output transistors in parallel, just different batches, in my JC-1 power amp. Yes, I have plenty of heatsink, etc. I'm amazed that we have even seen a problem. If your emitter resistors are large enough, and the betas are not too different, it will probably work.

BJT output stage emitter resistors

The optimum drop across a voltage driven complementary bipolar output stage is 15-25mV across each emitter resistor. Somewhat more or less will still work.

This is derived from an article published by HP in about 1971. This is the best area of operation for lowest distortion in the transition between class A and AB.

Folks you are just going to have to 'trust me' on this. It comes from a computer simulation that gives a range of possible values. The optimum value is between r(e) and r(e)/2, R(e)=1/G(m) of the transistor.

This won't work for FETS.

Does output transistors ages?

I hate to throw everyone off, BUT if you run a transistor very hot, it will wear out in a relatively short time. It is most probably migration of the metalization used to make the connection to the outside. I have experienced this, in a real design.

Class A with AB

Heavy class AB is best in my opinion, and I design the majority of my power amps that way. There 'can' be a problem with the crossover from class A to class B. I first found this out 35 years ago, when I used too large of value emitter resistors in my first personal amp design. Heck, I thought that 1 ohm with a .5A standing current was OK, but NOOOOO! Actually I should have used a .05 ohm resistor or so, for best transition. This is because an emitter follower, when voltage driven, prefers a small emitter resistor that equates to about .015-.025V when the amp is idling. The problem with this solution, is that thermal runaway of the output stage is possible, UNLESS you do your bias thermal compensation right. Still, it is possible to get an almost perfect transition from class A to class AB if you do it right.

Emitter resistors in output stage

This is the situation: YOU MUST USE A SMALL VALUE EMITTER RESISTOR IN EACH OUTPUT EMITTER. Yes, if you are an expert, perhaps you can design advanced thermal/ DC output current sensing and avoid the emitter resistors, BUT you better know what you are doing. What happens is that the Vbe junction heats up in operation and draws more average current, then this extra current heats up the Vbe junction even more, and even more current flows. Sooner or later, flash/bam almost all circuits blow up. Please keep this in mind. There is no point in using a collector resistor, except a high value one (or 2) to provide feedback to the input, and it won't have any significant current flowing through it.

Mosfet driver stage

I use a mos fet driver stage, because it presents a high input impedance to the pre-driver stage. This avoids modulation of the open loop gain by changes in the impedance of the loudspeaker load.

>But in your design I noticed that the sources of the driver mosfets is attached to the outputs by 22ohm resistor. So the driver is driving the speaker also?
This is a topological artifact that was developed by Parasound engineers for the HCA-2200. I have never changed it to a more orthodox format. It seems to work OK, but I have never tried to compare it in the same circuit.

High Power Output Transistor

Instead of just guessing, or giving opinions, I suggest that you LOOK UP the different output transistors and get the engineering data sheet on each one. Then, plot on a separate piece of graph paper, the BETA LINEARITY with current, on a LINEAR vs LINEAR plot, which is different from the LOG vs LOG plot that is normally presented with the engineering data sheet. Just overlay the different transistors with a common beta at one specific current, then note which are more linear. F(t) or intrinsic speed is important too! Third, plot the SAFE AREA of each device at some common time constant, like 10ms. This way, you find the most 'rugged' devices when you must have it, in order to keep the amp from blowing up.

Power Mosfets

At first, power mosfets looked like audio nirvana. They had high input impedance, a more linear G(m), and were free from the dreaded 2'nd breakdown that made transistors fail. When we actually tried them, however, we found big problems. First, the Hitachi 'lateral' power mosfets, while rugged and high voltage, had LOW G(m) and low peak output current.
The American 'vertical' fets had lots of G(m), fairly high voltage and current, BUT they were prone to breakdown. You can't use them at anywhere near their rated voltages, except as switches. And over the years, bipolar transistors got faster and more rugged. In the last 5 years or so, power mosfets have actually gotten worse. Many parts that are really superior, have been discontinued. Still, I will use selected mosfets, before I use bipolars, where I can, because they tend to give lower amounts of higher order distortion in my designs.

Class A

Class A is best, on all accounts, but it is not practical with very large power outputs. Mostly because it heats the room too much when operating big amps. The actual power spec that we normally relate to is AVERAGE POWER, that is derived from the RMS Voltage.

The only schematic that I have ever published on a power amp was in 'The Audio Amateur' in 1981. That was already a 5-10 year old design at the time, so it doesn't have servos, etc that I would use today. It would be better to get an old Parasound 2200 and mod it up.

Is P-channel FETs more linear?

I doubt it. Make sure that the OUTPUT of the device under test is the same with N and P. P channel devices usually have lower Gm, SO, output is lower with a given input. This makes distortion appear lower.

Base stopper in the output stage

The reason for a base resistor is to add a POSITIVE resistance to the base of the transistor. This counteracts the NEGATIVE resistance that can appear at the base of a transistor follower with a cap load. This is graduate school stuff. Many of us learn it the hard way. Too high a resistance makes parallel transistor matching difficult and slows down the output stage. Too little can cause random oscillations. 10 ohms is a good starting place.

Output protection

Output protection is really tricky. Almost all power relays are marginally lousy. Output fuses suck! However, rail fuses, or even better: dual DC circuit breakers work OK. E-I active protection usually causes big problems as well.
Parasound uses output protection relays because we have to. Our amps are usually pretty powerful with lots of peak current. Without protection, too many broken speakers would happen because of audiophile mistakes as well as an internal amp problem, and we would be held responsible.

Class B and AB idle current

It has been found by myself, that running as much current as possible without overheating is usually the best. If you want to get the best transisiton between class A and class B, which almost always is necessary, then with output bipolar transistors, drop 15mV-22mV across any emitter resistors. This is a good first approximation.

Emitter resistors in multiple BJT output stage

In general you should have 12-22mV across each emitter resistor. Going too low will cause problems, not only in thermal instability, BUT with the transistion between class A and class AB. Going too high will also cause added distortion in the transition between class A and class AB.

Output filter

I would put a 5-10 ohm resistor in parallel with the output inductor (2uH). Everybody does, for good reason.

Protecting output FETs

The situation to me seems that it would be a good idea to protect gate of the mosfet with a zener. It is just good engineering practice to do so.

Now, the stopper resistor is another question. I ALWAYS use a stopper resistor with my mosfets, and sometimes even in the base of any bipolar output transistors. This is also just good engineering practice, but for certain RF designs, this stopper resistor could be a problem. There are more exotic alternatives, but a resistor is pretty darn cheap and easy to do.

For the record, it is more difficult for a fet follower to develop enough gate voltage to break it, because the input follows the output. BUT, if the output was at very low impedance (like my WATT 1's are at 2KHz) and the input was very high, then the gate could be breached in an instant. Probable? Maybe not, but it is prudent to cover all bases.

Cziklai feedback pair vs darlington connection

It is difficult to separate 'feedback' from 'degeneration' as both terms are used.
The best reference of the practical difference between a complementary feedback pair and a darlington follower comes from 'Analysis and Design of Analog Intergrated Circuits' by Paul Grey/ Robert Meyer' pp 398-399.
"The major problem with the configuration 'complementary feedback pair' is the POTENTIAL INSTABILITY of the local feedback loop formed ... particularly with capacitive loads." I said it before and I say it again: The complementary feedback pair is LESS potentially free from oscillation, all else being equal.

Loading power amp

For some reason, there is confusion about loading power amps. In theory, all solid state amps can run without a load. With tube amps, the output transformer apparently reflects the load in some way to the amp, and this can cause problems. IF you are still concerned, just put a 100 ohm 2W resistor or even more across the output. It won't get too hot, but it will potentially damp the output stage, if that were at all necessary.

3.3 Feedback network

Eliminating capacitor in amplifier series feedback loop

Almost everyone puts an electrolytic in the feedback loop. The reason is that it is cheap and effective. Servos, properly designed, are better. Some extreme friends of mine, choose to direct couple, without servos, and live with the voltage drift on the output. This is not easy, most of the time.

Servo

I find that servos work for me, but you can do what you want to.
As far as hum pick-up is concerned, this is a quote from a 'Stereophile' review of the Vendetta SCP-2 phono stage using an Ortofon 3000 input stage, that has about the same output as your Ortofon cartridge. "When no disc was playing, there was a very faint, muted hiss and not a trace of hum." So much for balanced input, because the SCP-2 is single ended. Gordon Holt June 1988 p.3 in 'Stereophile archives.

Servo

it is important to find novel ways to steer the output so that it has no DC offset. Personally, I believe in servos. BUT, it seems important to me that the servo is only significantly active at VERY LOW frequencies, and not as a low frequency roll-off. Years ago, I actually tried to make a servo also act as a high pass filter (ie 50Hz). This was not so good. It appears to be better to just use a quality cap to do the same thing. Usually, it is best to minimize 'global' feedback. Probably, feedback pairs are better, but not perfect. In any case, the most linear circuit, (without feedback) is usually best.

Servo

Usually we put input limiting on a servo to keep it from 'latching up' This can happen because the drive voltage to the servo 'can' exceed the power supply voltage. Diodes are a crude fix, but series reversed polarity zeners, maybe +/- 6.2 V or more might be better. Diodes will always conduct a little and with a high Z source, can conduct enough to create a non-linearity. Zeners will not come on as easily until a certain threshold is reached.

Servos get a bad rap. I don't claim that they ever can be 'perfect' but usually they are designed to respond too quickly in order to give a fast turn-on, and then tend to respond to the asymmetry in any audio waveform coming from the source. We can be pretty sure that DC is NOT part of the actual audio signal, but asymmetric waveforms are common. This creates a DC like component that can effect servos big time. The audio signal cannot maintain this DC component for long, but if the servos are quick to respond, then they will start to significantly put out a 'correction signal' that will change the waveform. This is not good.

It would be better to use a coupling cap to roll off this DC component if it was necessary to do so.

3.4 Power supply and regulators

Dual bridges

Dual bridges allow that the power transformer does not have to be connected to the ground to create a center tap.

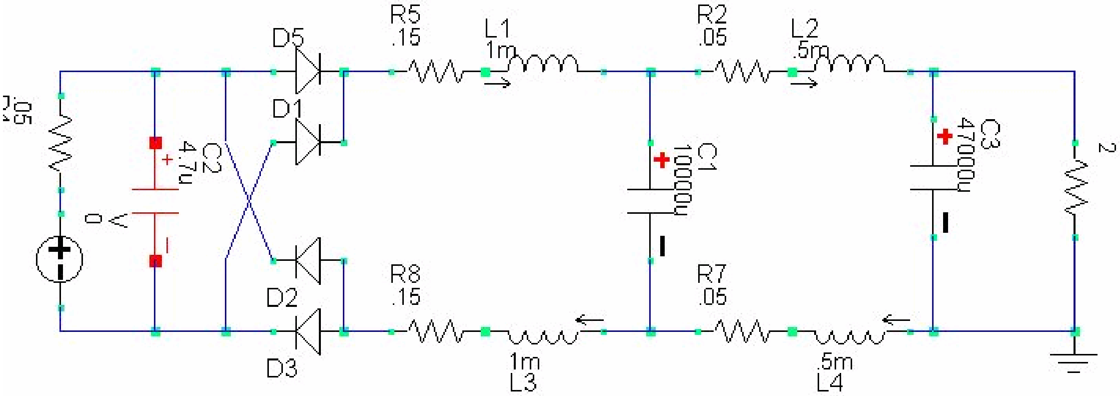
Multiple bridges can give a real improvement in audio quality. That is why many designers now use this technique. First, all power transformers have capacitance between the input and the output windings. Yes, you can pay extra for a special isolation tranformer, but generally, the available transformers are fairly lousy, especially toroids. By not connecting a center tap to the ground, you create another layer of isolation between the power line and the internal ground.
About 15 years ago, I was also suspicious of dual rectifiers. However, while I made my Vendetta preamps, I decided to change over. I found that a potential ground loop had been broken by using the dual rectifiers.
This is one of the serious problems with Spice emulation, and little or no hands-on experience. The actual operating conditions are seldom re-created
in the Spice emulation, and potential advantages are overlooked.
There is one warning about using dual bridges. This is because the +/- loads have to be nearly exact, or the DC voltages will drift. A center tap usually drains off any difference.

No one wants to use more diodes than necessary. Good diodes are expensive and often are not available as bridges. This means more time and effort is necessary to make a proper diode bridge, and even more cost and complication to make a dual diode bridge, BUT we do it, when we want to do it best for preamps, especially.
The basic idea is to make as isolated an AC-DC power supply as possible. This includes a good power transformer with high isolation, high speed diodes dual diode bridge power supply, common mode chokes, and good electrolytic caps. Anything less is a compromise.

We tend to use 'soft recovery' high speed diodes. If you use a current probe, you can easily see what slow diodes do.
I resisted this for several years. It, at first, made little or no sense to me as well.
My Vendetta Research power supplies went through many small, but important changes over 20 years. First, I just used pre-packaged +/-15V supplies, just like Mark Levinson did even 10 years earlier.
Then, about 20 years ago, I started to build my own.
First, I used a single full wave rectifier bridge, and a few thousand microfarads load capacitance.
Then, about 5 years later, I tried dual bridges, just for a special project. We found that we could hear the difference between this new power supply and the standard power supply that we replaced. In fact, J Gordon Holt, ordered one from me. Then I was reminded by

my associate that I was only using a single bridge, so we changed, and this became our 'A' modification. I found that it was easier to remove any potential ground loops with these new power supplies.
Then, a friend of mine tried high speed diodes, and published his results in 'The Audio Amateur' in the early '90's. Well, I had stopped making Vendetta products, but I had a friend (now business partner) Bob Crump, who replaced the dual diode bridges with discrete high speed devices. Guess what, it improved things, even though the power supply was remotely located. Apparently, stuff gets into the grounding. I then made some measurements with a current probe and was I surprised! Wow! The standard diodes spit out all kinds of stuff. Check it out.
Today, I use high isolation transformers, high speed, soft recovery diodes, with dual bridge supplies in any preamp that I make, as well as quality caps, common mode chokes and triple active filtering.
Remember, we are trying to make the equivalent of a battery, without the problems and cost of using batteries. This is not that easy.

Opinion



You are getting me worried! I design power supplies like this.

Battery power

Batteries are best, all else being equal. However, batteries are expensive to maintain and replace. Trust me on this, they just don't last, and you have to replace them, far earlier than any AC supply. This is a hassle, expensive, and time consuming as well. Think about it. The battery develops problems, usually a cell short. The voltage drops below normal. You have to replace the battery, but you also have to turn the old one in for recycling.

Power supply capacitance

Use as much power supply capacitance as is reasonable for you. It is not because of hum or noise, but the impedance of the return path of the loudspeaker to the power supply. We have found that 20,000 uF and more is necessary, per channel, to make the bass sound right.

Measuring rectifiers

Look at the CURRENT waveform, rather than the voltage waveform. You need a current probe, current tranformer, or something special, BUT you should see more than what you are seeing. Hi speed, soft recovery diodes are necessary for any of my hi end products.

PS for class AB vs class A

For the record, Class A amps need a BIG power supply, BUT is it fairly continuous with current at different audio levels, so absolute regulation is not so important. Class A-B, B amps can, in theory, use a smaller power supply, BUT the supply will have very different current demands with different audio levels. IF the bias regulation in the amp is not buffered well enough from the supply, then problems with the power supply can actually change the bias setting.

W.Jung regulator

I personally don't use the Jung 'super-regulator' in any of my products, but I respect everything that Walt does in this area. Look for more, in future. However, it is very important to make the best AC-DC filtering system possible, in order that the regulator is taken out of the sonic imprint. Trust me, it is NOT easy, except with batteries.

If it was up to me to 'improve' this regulator, I would try to use a video IC for the gain. This would give, all else being equal, faster response, more linear feedback control, and about the same noise.

Stick with the AD825.

Regulator improvement

When 'optimizing' a regulator, several factors must be thought through.
1. What do you want it to do?
2. How noisy is it?
3. What is its practical output impedance? Over frequency?
4.What is its transient response, both to load changes, and source changes?
There may be many other factors that I cannot think of just now.
When you change op amps, you change many of these factors.
Please keep this in mind.

Feedback regulated power supplies

However, with feedback regulated power supplies, it is best NOT to use a really high Q, (good) cap at the output. The reason is that the output looks like a synthetic inductor, BECAUSE the output impedance change with frequency will act exactly like one. Therefore, if you put a really good cap at the output, the Q of the resonant circuit formed will make it ring. This usually is seen as an extra noise, but it could show itself in other ways. SO, either you put a resistor in series with a good cap, or you use a lossy cap. Take your choice.

Fuses impact on sonics of a supply

I just received an E-mail from an associate concerning fuses. Apparently, they changed the sound in one of my designs. I also spoke with a German guy who made exotic audio fuses, at CES, and he made sense to me. They are pretty expensive though, and I would think that other things should come first.

Regulator

Zener referenced regulators are noisy. Cap referenced regulators have poor DC stability. A combination of the two, or even better: One zener referenced regulator (IC can be OK) and then a cap multiplier to smooth noise and give good high frequency performance is even better. Be careful NOT to share cap multipliers with different channels. They can xtalk. Best to use a separate cap follower with each circuit or at least, each channel.

That's the way that I have done it for decades. It works!

Voltage references and current sources

I think that you could change the green led to a red one and get lower effective voltage drop. Heck, you could use 1 diode instead of an led. Then a FET current source would be really more important. You should use current source fets if you can, and are not expert. They are already selected and optimized at the rated current. The Vishay j202 seems to be a pretty good deal, if it is like the j502 in performance.
Don't expect VERY high effective impedances with fets, unless they are long gate devices. This would be unusual with cheaper fets. Cascode is recommended in many cases.

If you NEED a current source, the j202 on reflection is actually a poor choice for a current source, UNLESS you know how to sort for current or calibrate it with a series resistor. This is beyond most people here, and it takes a certain amount of time and a large number of parts. There are better parts with a lower Vgs, such as the 2SK170 Gr that would work OK, once selected out to the right current value of Idss.

First of all: Do we just want to use a fet as a current source from the supply input? If we do, then we have to use a low Vp device. That means, high Gm. It will have marginal output impedance and it will amplify its own noise, but it can be made to work.
Or do we want to use a fet as a current source to bias a LED that is connected to the supply, while the fet current source is connected to ground? Then, a relatively high voltage 40-50V part, usually a long gate type, like the J202 is recommended, as it will allow relatively high voltage across it, and because it has low Gm, will not amplify its own noise as much. A current source fet will work as well, but they are fairly noisy, and

relatively expensive. Their greatest advantage is that you know the current they will pass, because they are selected at the factory to have a specific current.
Effective impedances above 20K are practical with the fets discussed here, but only a cascode will get you there, for sure. Watch out for short gate devices (high Gm) They sometimes look like a triode, rather than a pentode.
I just checked a j232 and it has a very high output impedance, much higher that I suspected that it would have.

Of course, we have a RATIO of 2 impedances that set the rejection. We have the effective impedance of the jfet, and the effective impedance of the LED.
Now think!
You can bust your tail to get the DC impedance of the jfet very high, BUT it will NOT hold with increasing frequency because of miller capacitance in the fet. The effective impedance will start falling off at 6dB/octave, sooner or later.
Now, the LED is a DIODE, so it changes its impedance directly with current. Therefore, increasing current through the LED will LOWER its impedance. The limit is the max operating current of the LED, and its intrinsic resistance, due to construction compromises. Usually, we would not usually bias the LED above about 5ma, and much below 1ma, as this will make the led noisy.
The j202, a popular and cheap device will work (barely), AND you don't even need an adjustable resistor, BECAUSE that would only force the LED to operate below 1ma, and that is bad for the LED.
Of course, we could use another part, like the j113, but why? First, it is a short gate, and has a more triode like characteristic than a long gate device like the j202, also it would HAVE to be adjusted to actually work. So why bother?
I hope that this helps.

For the record, GREEN LEDS are OK, but RED LED's would give you more 'headroom' if you needed it. This was my point. I often use green LED's myself, but then I usually have plenty of extra voltage at the input.

4 Design examples
4.1 JC1 amplifier

JC1 measurements
In my JC-1, the same measurement was made at over 600W into 4 ohms. Is this listening level? No, but many good amps take extraordinary measurement equipment and techniques to get the higher order harmonics measurable on traditional test equipment. For example, with 20W into 4 ohms into a JC-1, I could measure a difference in the 5th harmonic between the high and low bias. About -115dB with low bias, better than -120dB at high bias. 'Stereophile' did not note this difference in their review. Still, I think that a 20W measurement is MUCH more important than at powers greater than 100W. Also folks, please consider the nature of the graphs presented. They are in dB! IF you looked at the graph on a linear scale, the higher order distortions would not be seen, except as a smudge at the bottom of the graph. This can distort the perception of the actual linearity of the design being tested.

I made 2 points about measuring amps. First, you have to measure the amp at its 'working level' to get an idea of what it will sound like from distortion measurements, and even that might not tell you enough.
This is not normally done by 'Stereophile'.
Secondly, it is important to NOTE that harmonic distortion levels are most often expressed in dB. This brings up the lower levels to almost match the higher levels. This is good for measurement, but bad for complete understanding.
Thirdly, I might also recommend a formula for 'weighting' the harmonics. This changes things again:
harmonic multiplier is equal to (n-1)!/2 This gives: 2nd=1/2, 3rd=1, 4th=3, 5th=12, etc.
This implies that higher order harmonics are VERY bad for audio quality.

4.2 JC2 amplifier

Well, the schematic is wrong. It was caught by me after this schematic was first published, in 1977, and was corrected by me in a LTE of 'The Audio Amateur' in the next issue or so.
Yes the polarity is inverted in the schematic.
You must reverse the connections to the emitter and base on both sides.

This is a pretty good circuit, but don't try to use the original parts. It is better to use the Toshiba 2sk389 and 2sj109 parts for the input. The second stage can be any darn good 1/2A-2A rated transistor complement. This design comes from 1973, more than 30 years ago, and we used what was available then. The mistake in the schematic came from Mark Levinson's file that he gave to me, and I had transcribed by a tech. I should have seen the problem before it was published, but I didn't. This circuit is defined as a transconductance amplifier, rather than an op amp. It has a slew rate of about 100V/us and a fairly high open loop bandwidth. It is the basis of all my designs for later preamps and power amps.
PS you HAVE to use a smaller resistor in series with the output emitters or it won't work. 20 ohms is on the original schematic.

Even with the leads switched, the second stage is still essentially a common emitter drive. I just added the alternate input lead to the emitter of the second stage because it seemed to be an OK thing to do. It does have some subtle advantages.
What you are thinking is that it becomes a common base second stage, and sometimes I do design circuits that way, but in this case the common emitter overwhelms the common base connection. However, if you put a large bypass cap at the base of each output transistor to ground, it would become common base driven and the circuit could practically only operate open loop. Both Chas Hansen and I design some of our latest circuits with a common base drive for the second stage. It is then called a folded cascode connection.

Now that we have resolved the schematic error, and the measured distortion, we should address WHY I bothered to design this circuit in the first place.

JC2

Well, the schematic is wrong. It was caught by me after this schematic was first published, in 1977, and was corrected by me in a LTE of 'The Audio Amateur' in the next issue or so.
Yes the polarity is inverted in the schematic.
You must reverse the connections to the emitter and base on both sides.

This is a pretty good circuit, but don't try to use the original parts. It is better to use the Toshiba 2sk389 and 2sj109 parts for the input. The second stage can be any darn good 1/2A-2A rated transistor complement. This design comes from 1973, more than 30 years ago, and we used what was available then. The mistake in the schematic came from Mark Levinson's file that he gave to me, and I had transcribed by a tech. I should have seen the problem before it was published, but I didn't. This circuit is defined as a transconductance amplifier, rather than an op amp. It has a slew rate of about 100V/us and a fairly high open loop bandwidth. It is the basis of all my designs for later preamps and power amps.
PS you HAVE to use a smaller resistor in series with the output emitters or it won't work. 20 ohms is on the original schematic.

Even with the leads switched, the second stage is still essentially a common emitter drive. I just added the alternate input lead to the emitter of the second stage because it seemed to be an OK thing to do. It does have some subtle advantages.
What you are thinking is that it becomes a common base second stage, and sometimes I do design circuits that way, but in this case the common emitter overwhelms the common base connection. However, if you put a large bypass cap at the base of each output transistor to ground, it would become common base driven and the circuit could practically only operate open loop. Both Chas Hansen and I design some of our latest circuits with a common base drive for the second stage. It is then called a folded cascode connection.

Op-amps

First, OP AMPS had been around for almost 10 years. In 1966, I worked with the UA702 and UA709 op amps, and the UA741 op amp in 1969. These parts were a god-send for minaturization and for servo control, but fairly lousy audio devices. In 1970, Harris Semi (then Radiation Inc) came out with a dielectrically isolated op amp with low noise (9nv/rtHz) 50 ma peak current, +/- 24V/us operation, and a slew-rate of +5/-2.5 V/us slew rate. Selected units could measure fairly low distortion as well. This seemed to be the answer to an audio designers needs, BUT once we used them, we found them not to be sonically as good as tubes.
What to do? Well I decided to build a discrete circuit with a fet input that had a minimum circuit thru-path, high open loop bandwidth, and as linear as possible operation for each device.

For line amp operation, the circuit that we have previously discussed worked for me. Both Mark Levinson and the Grateful Dead used this circuit as line drivers for several years.
For higher closed loop gain needs, another circuit design was necessary. Then, the op amp configuration works better.

Let's go on, if we can, as to WHY we would want to build simple circuits, rather than use complicated thru-paths? Now, when I mean 'simple' I don't mean crude, or elementary. Push pull is OK, so is 4 quadrant operation so that you can have both balanced inputs and balanced outputs. This can be useful, even when using only one output, as a phase inverter for generating absolute polarity with different software.
Also, think about distortion and what the harmonics look like.
And finally, think about high open loop bandwidth, which is difficult, but not impossible with OP Amps. Why would we want high open loop bandwidth?

It is interesting that I made a modified version of this design to make the Grateful Dead Line driver, that has to send the stereo signal from the mix board in the audience to the stage, 100 ft or more. This is probably the nastiest load that anyone here would ever encounter.
Actually transconductance amps are MORE STABLE than op amps. This is because, instead of ringing, they become more compensated by the load capacitance.
An op amp would have a follower of some kind. In this case, the cap load is buffered from the compensation, and this creates a separate second rolloff of the high frequencies. This is what causes ringing.
The synthetic inductance is another issue, and an interesting one. I suspect that you could build a 'pathological' transconductance amp that had low gain-bandwidth and low open loop bandwidth that could be problematic. This particular design is high gain-bandwidth and high open loop bandwidth, so it may not have as much problem with synthetic inductance. However, it is an interesting question.

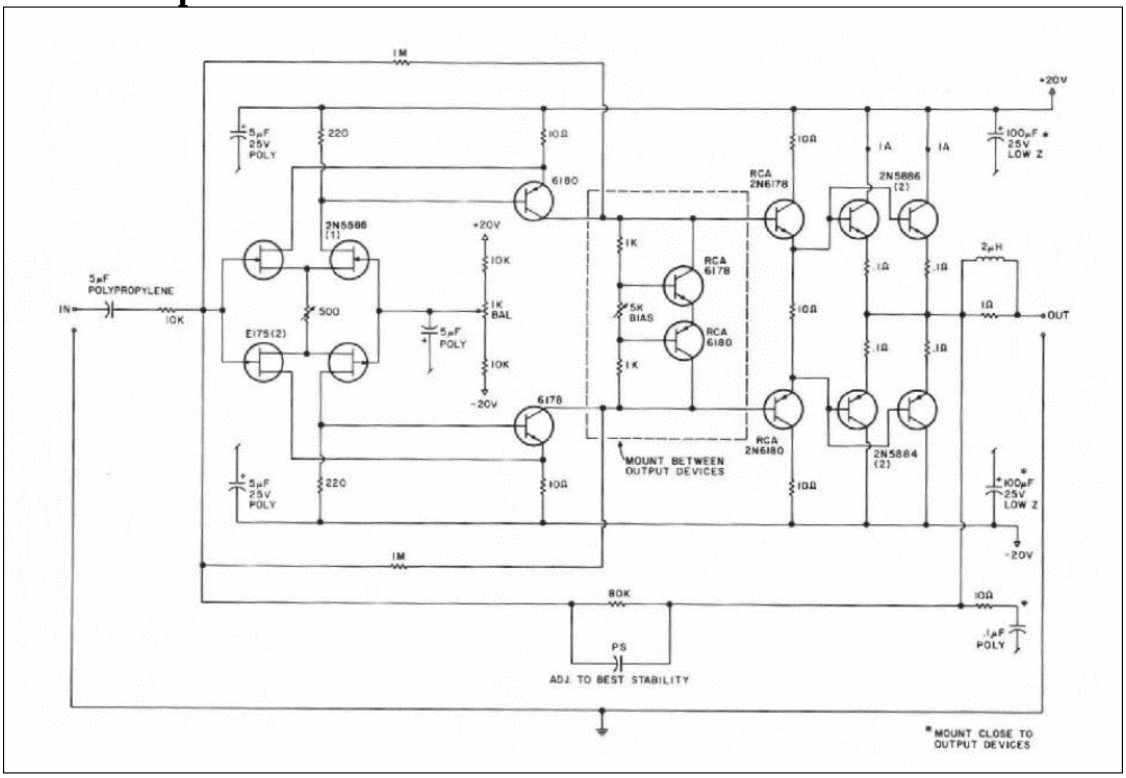
Ron Wickersham, Bear and I had the responsibility for the components of the system. There were others as well. Jon Meyer was a personal friend to me, but he did not work with this system. Later, Jon Meyer and I went to Switzerland to work on another project, and this system was completed by Ron Wickersham.

Another small fact. The circuit that we were originally talking about on this thread that came from the JC-2, was originally designed for the big 'Wall of Sound' system used by the Grateful Dead. Mark adopted it for the JC-2 because of its success with the Grateful Dead in sounding better than selected existing op-amps.

JC-2

The JC-2 is not an EASY design to make. Many of the original parts are obsolete. The circuit topologies are still viable, and almost typical, today. Thirty years ago, this was an advanced design. Elso has not been productive in improving this design, and while I respect his opinion, I find errors in his estimate of the sound quality potential of this design. Done right, it can still keep up with most anything today. There are some oversights, however, that we made in 1973-74, that need to be fixed. We attempted an upgrade to existing units in an article a few years ago in 'Positive Feedback' written by Chuck Hollander. It might be useful to find this article on the internet.

4.3 JC3 amplifier



That design is correct in principle, but the design is over 25 years old, and the parts are obsolete. The basic design can be modified with FET's just about everywhere. I recommend Fairchild Mosfets, as they will be better than IR in this application. Jfets can be 2sk389 and j109.
This design is somewhat limited to inverting operation, but can be more useful with a servo added to control the offset.

This is a dual feedback arrangement. Especially in the early days, it seemed to be useful to reduce the amount of global negative feedback, because of potential TIM. This technique lowers the over global negative feedback to the ratio of the internal feedback resistors and the overall feedback resistor. This should be about 14-20dB. The reason for this is that the drive impedance to the output stage is lowered and this removes the dependence of very linear beta in the output stage. Another way of looking at this is that the pre-driver stage (6178-6180) generates a very high drive impedance. The darlington output stage reflects back from the speaker impedance another fairly high impedance, which is essentially 2*B(1)*B(2)*R(L). It is the relative levels of these two impedances that implies whether the output stage is voltage or current controlled. Lowering the drive

impedance to the output stage makes the output stage into a voltage follower, rather than a beta multiplier (which is more nonlinear).
With fets in the circuit, this is unnecessary, as the gate is never current controlled.

A transistor output stage can be thought to be controlled in two ways. It can either be a BETA MULTIPLIER or a VOLTAGE FOLLOWER. What matters is the source impedance that the output stage is driven with. If it is driven with a HIGH impedance, then the output stage will behave as a Beta Multiplier and will therefore be VERY dependent on BETA.
If, on the other hand the drive to the output stage is LOW impedance, then the transistor output stage will behave as a Voltage Follower to the drive voltage. Beta nonlinearity will then be less important.
The extra feedback loop reduces the drive impedance to the transistor output stage, so it should become more linear, and also be less sensitive to load variation on overall linearity.
I left out one important word in my previous statement:
It should have said "removes the dependence of HAVING a very linear beta in the output stage"
I hope you can understand what the extra feedback does better now.

I used the inverted mode in the JC-3 in order to use a film cap rather than an electrolytic cap in the feedback loop. Also, it removes common mode distortion generation, because the input stage is not cascoded.

Common mode distortion is caused by a common mode input signal appearing at the amp inputs. This is both + and - inputs. When you have a non-inverting amp with low voltage gain, the input signal and the feedback ride up and down in common mode, with the drive signal. If you do NOT cascode, then you get more distortion. This can be also seen with analog IC's, at least early devices that were not cascoded on the input.
The inverting input FORCES the - input to 0 volts at all times (for all practical purposes. The + input is already at 0 volts, so no common mode signal is produced and therefore no common mode distortion.
The JC-3 is an OLD design about 30 years old. It will still work, but it is not a good example of modern designs, except for the internal topology.

It is essentially a JC-2 line driver, coupled with a darlington output stage. You are correct that the dominant pole is created by the miller capacitance of the driver stage transistors. This design, as it stands, can do about 100V/us and should have an open loop bandwidth of about 20KHz (as I remember). This was to fit the critereon set by Matti Otala at the time.
Of course, newer, faster output transistors would make this amp faster, the initial parts had an F(t) of 4 MHz. We can get 10 times this, today.
The RCA devices initially used were actually better at the time than most equivalently rated transistors, because they had less distortion due to their internal construction, but I doubt that they are available today.

This design topology is what is used in all the Parasound amp designs that I am associated with.
Because of faster output transistors, we have eliminated the output coil and still get 100+ V/us slew rate.
This design, with improved output transistors, would work with unity gain, but it seems to be overdesign for just a buffer.

JC1
The JC-1 is just bigger, and more complex. It has to swing +/- 90V rails. I cascode the input stage with fets, the driver stage is composed of 200V fets, and the 'darlington' driver is now fets as well. All that is left are the bipolar output transistors. By the way, be sure to add 10 ohms to the base of any output transistor used. We found that with ring emitter devices (high speed) that this is important. The 10 ohms adds a real resistance that cancels any negative resistance generated by the combined phase shift of the base emitter junction and any capacitive load.
Again, the reason for the inverting input was to reduce COMMON MODE distortion to a minimum. It is not necessary IF you CASCODE the input stage. All modern designs are non-inverting or balanced input.

JC-3 improvement

If I were to improve on this design, I would use all FET's. The 2SK389 and the 2SJ109 are excellent input devices. I would also servo the design to remove the input cap and the offset adjust. Just a suggestion.

J76 is a good part, but it looks like a triode. The K213 looks like a pentode. The only solution is to cascode the second stage to reduce distortion to acceptable levels. This is a serious trade-off.

The amp in the schematic is class A, because it has high standing current (without signal). Many suggestions have been to compromise the original amp with inferior output devices and LOWER standing current. This is bad! Why bother with an inferior design? There are newer active devices that can replace every device, that are superior to what was put in originally, about 30 years ago. I should hope so! Fets could replace virtually all of the bipolar devices, and even the driver stage could be removed, because it would not be necessary. Let's see if we can get some more responses that make sense, in this example.

learn about the devices that you are recommending, before putting them forward. For example, the MJ15003-4 combination has 1/2 the beta linearity and 1/2 the gain-bandwidth of my original 5884-5886 combination. There are MUCH better parts available today. The Hitachi K213-J76 series is flawed in a different way. You MUST cascode them to make them work reasonably well. The output conductance is completely dissimilar in the two devices. This leads to big-time 2'nd harmonic.
Some of the other circuit designs shown on this thread will work, but are not optimum, either to build or for other reasons. I hope that this helps.

I also used MJ15003's and 4's in a bridged 250W power amp that I designed for Gale in 1976. We started with the 5884-6's in the prototype, then we changed the parts to the MJ's. Well, the distortion doubled and the slew dropped in half. How would you know that? Today, we have LOT'S of parts that are better than either device in every way. We just have to find them, that's all.

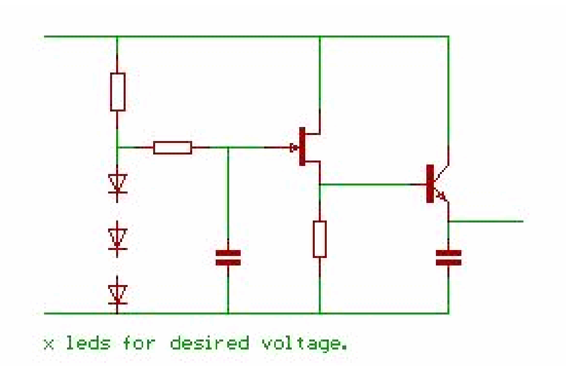
4.4 CTC Blowtorch amplifier

Power supply

Let's not start 'second-guessing' the power supply as well!
I think of the power supply as doing several functions. First, it must separate the channels from each other. IF you don't use a separate regulator for each channel, you will have extra crosstalk on this design. Therefore, each circuit board and each stage (in my design) has a series open loop regulator. There are 8 series regulators in the CTC preamp in the picture.
The other chassis contains 4 regulators Two series, and two shunt. It also has a passive PI network to block RFI and keep it off the ground return.
Now, after almost 40 years, why have I done things this way?
Well, 40 years ago, usually the ONLY regulator would be a single series type for the entire preamp. This was true with AMPEX audio pro recorders, or Dynakits.
Mark Levinson introduced the first +/-15V potted module active linear regulator to audio in the LMP-2 preamp. I used this approach, BUT I had to quiet the inherent noise of the regulator (zener stabilized remember) with an active low noise cap multiplier. This seemed to work great for years, BUT I realized that it introduced considerable (xtalk) between the two channels, because of the relatively high output impedance of the open loop regulator, especially because Mark added a 2 ohm resistor to the output for current protection.
The JC-2 had a problem with imaging. It was great for mono, but stereo was not as good as some other products. I traced this partially to the power supply buffer.
For Vendetta Research, I had to use a more sophisticated approach, in order to make it extremely low noise, and not contribute a sonic 'character' to the circuit. I found that ANY aluminum or tantalum cap that was used directly across the input circuit power supply would change the sound. Therefore, I had to find a quality film cap that would work OK. The caps are prominent in the picture of the CTC. This also forced me to use all fet followers, and remove any bipolar devices. Does anyone know why?

Noise in BJT

Bipolar transistors have 3 major noise generators. The one that I am concerned about here is the shot noise generated by current flow across the base-emitter junction. IF the base-emitter junction is NOT effectively shorted at audio frequencies, then the base current shot noise will dominate and make the regulator noisy. IF you cannot effectively short circuit the base-emitter junction at audio frequencies by using large value caps from both emitter to ground and also from base to ground, then we have a problem. A fet is different. It doesn't have any significant current flow at the input, so it doesn't get noisy, IF there is not a large capacitor at the output of the 'open loop' follower regulator. Once I heard differences in electrolytic caps in my Vendetta input circuit (thanks, Peter Morcrieff for showing this to me) over 20 years ago, I had to design them out. The 0.1 polystyrene cap on the output is not doing much at audio frequencies, and it sounds good as well, for some reason.



Personally I like your design, AND you avoided many oversights often made by others. Your approach is truly low noise. The output transistor is driven by low impedance and the output is 'shorted' at audio by the large cap at the output. Your DC stability is OK, but could be improved, IF you felt it necessary by using a current source instead of a resistor, but it will still work well enough and it will be very quiet at audio frequencies.
Now, why don't I do this? Well, as I said in a previous comment, I can hear the 'sound contribution' of the output electrolytic cap when it powered my folded cascode circuit. As an engineer, I would not have thought it possible, but I heard it in a direct AB test with a large value film cap, that sounded better. This forced me to use another approach, still very similar to yours:
I just used the FET driver directly and removed the output transistor and the large output cap. Now what is the tradeoff? Well the output impedance WILL go up, BUT if I use a regulator for each individual circuit that is running class A at all times, then the 5-20 ohm output impedance might be OK, because the change in Gm will be very small, because the power supply current load will be essentially constant. Then I don't need the final large cap. There won't be any added X-talk either, because each channel uses separate regulators, in fact each gain stage has its own regulators. That is why I use so many regulators in my circuits. In the picture of the CTC there are 8 visible, and 12 total, in that box alone. With the power supply box added, there are a total of 28 active regulators to drive the audio circuits in this box.
Now, what about DC regulation? Well in the actual power supply box, I use conventional 3 terminal regulators, to remove hum and ultra low frequency (breathing) of the power line. I have found that 2 regulators in series is better than one 'super' regulator. I hope that this makes sense to you.

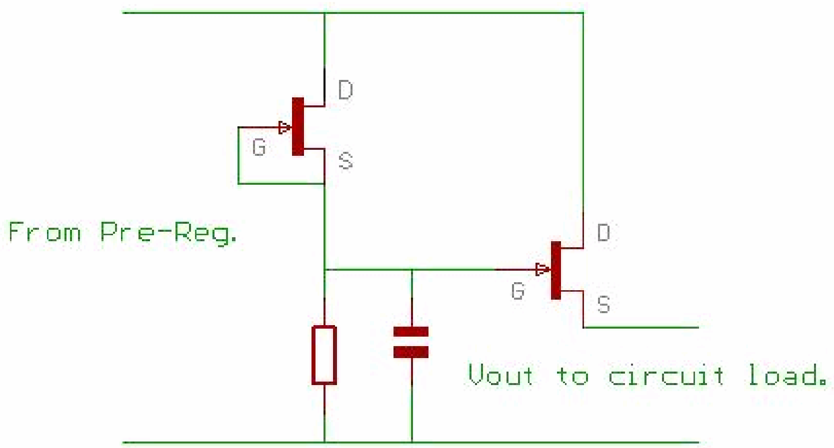
Capacitors

There are many imperfections in capacitors. DA is related to the material in the dielectric. Aluminum electrolytic caps tend to have the worst DA. Dissipation factor is only slightly related to DA, but is most concerned with heating of the cap, due to its losses. This

includes DA, but also connection resistance and lead resistance. DF is important in switching supplies so that the filter cap does not overheat and explode.
ESR is again related, but is mostly concerned with the 'short-circuit' aspects of the cap, and how well it will pass signals without any drop across it.
These are approximations, and you all can quibble with me about them, but it is important to understand that they are not all the same thing.
DA was found to be important, first, in analog computers, popular in the 1950's. It was known to effect calculations, and was difficult to compensate for. It is also very important in sample and hold circuit operations.
Tantalum and ceramic caps can have lots of nonlinear distortion as well. Aluminum caps are better, but not perfect in this respect.

'Smear' is a pretty good description of DA. I once used the term 'echo' for DA effects, more than 20 years ago in a LTE to 'Wireless World' or 'Hi Fi news'. I never heard the end of it from Doug Self, etc. , but 'echo' is a good first approximation. It is signal AFTER the original input has gone, and there should be only silence.

I recommend looking at Photo #6 of a 10uf aluminum cap compared to an 8 uf polypropylene cap differentially subtracted with a (nominal) 50k load on both caps in the article: 'A Real-Time Signal Test for Capacitor Quality'. This article can be found on Walt Jung's website.
With a differential test, one can see that the signal difference between the caps lingers many times longer than the original test signal.
It looks like 'smearing' to me, but then, what do I know?

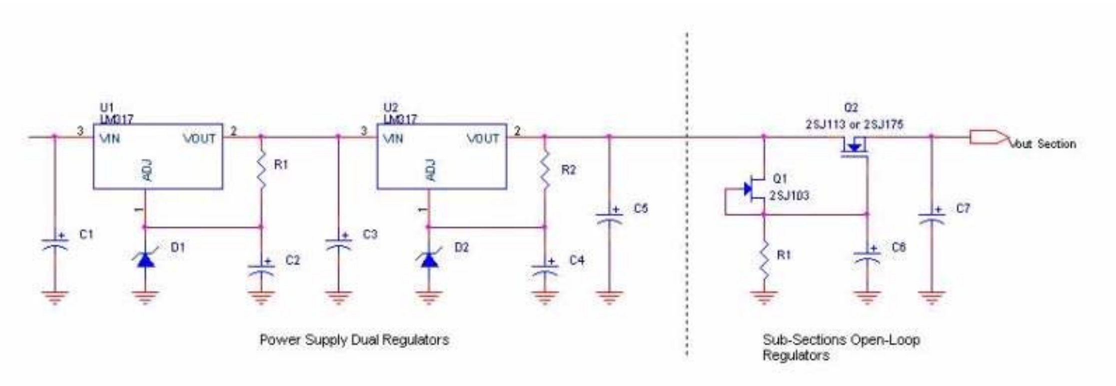


You have discovered the topology of the input regulators of the phono input board. Good work!
This circuit is both voltage stable and low noise.
Now why jfets on the input board and mosfets everywhere else/ Well, mosfets are easier to bias and are more rugged than jfets. However, they are also more noisy.

So, I use jfets for the input board, Hitachi 1/2 amp mosfets for the second stage phono, (they are quieter than most mosfets) and Harris 1A mosfets for the line amp regulators (noisy, but easy to buy).

I might also comment that the current source (fet) and resistor load is a Norton equivalent to a zener (or LED stack). It sets a stable voltage, but you must use a low gm low noise fet so that you do not amplify the fet's noise significantly. Also, you have to select the fet and match it with an appropriate value resistor to get a repeatable voltage. This makes it harder to use than a zener, but it is much quieter.

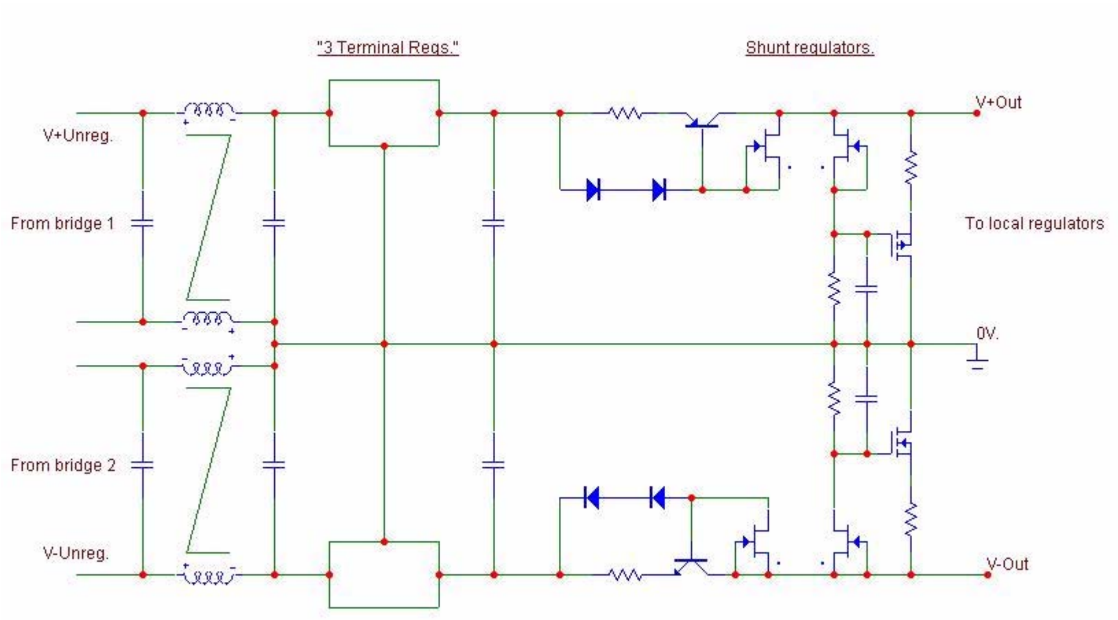
j203 is my first choice, and what I used in my Vendetta input stage. Unfortunately, they have to be sorted to .1ma in Idss and then I have to match the selected current fet with a range of 1% resistors to get the voltage right. (and people wonder why my best products cost so much) The follower is a J113 or a J175, as these are also low noise, BUT have higher transconductance for better voltage regulation. The best parts for the followers would be Toshiba low noise complementary parts, but I didn't want to 'waste' them on this application. They have almost always been hard to get, and better used in the preamp circuits directly.
The first stage regulator is not just a cap multiplier, but a Norton equivalent voltage stabilized follower, with a cap added for lower noise. Since this cap does NOT have to 'dance with the audio' so to speak, it can be just a good electrolytic cap. It doesn't even see an audio signal at any time.
You are correct that the higher level regulators are cap multipliers, as the absolute voltage is stabilized by the 3 terminal zener regulated feedback regulator.



Close, but the second regulator is an open loop shunt regulator.
Is the shunt device BJT based or Mosfet based for the shunt regulator ?
Vertical Mosfet.

It is the idea of having to use so many regulators, each of which does a different job. The shunt regulator tends to capture any high frequency transient that might get through or even be generated by the first regulator. There are several equally good combinations that should work well.
What is more important is the passive devices and configuration ahead of the first regulator. It is designed as a pi network to remove potential ground contamination from

RFI, etc. Soft recovery diodes are used as well, without using a center tap on the secondary of the dual coil power transformer. You now see examples of this in many designs. Years ago, it was considered 'excessive' and it is still somewhat difficult to implement.



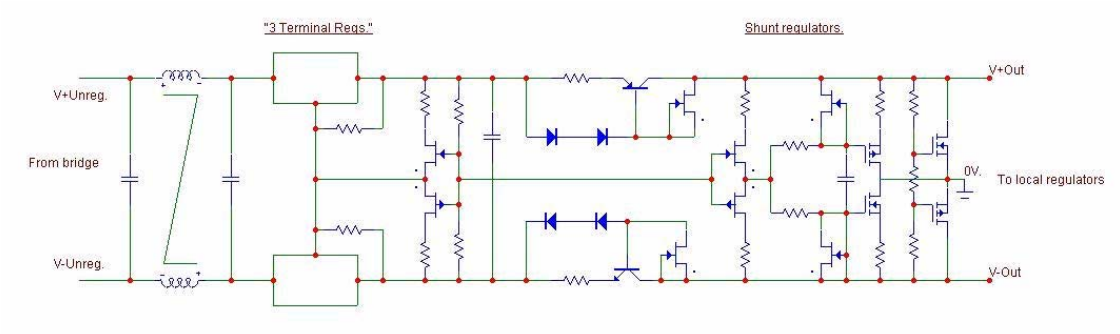
pretty darn good shunt regulator. I learned a trick or two from this schematic. I also like your CM choke. I use one in exactly the same place.
In fact, the left half is almost exactly what my schematic looks like, including the input power cap configuration.
As far as shaft extenders go, they can be a good choice. We are using them in the Parasound JC-2 preamp that was first introduced at the CES show to extend the pot shafts. I don't know what to say about switched grounds, but they could be a good idea.

I would guess that suspending those heavy switches and pots in mid-air over the line amps would have cut down some wiring.

Actually I may, at most, use 3 regulators in series. One is feedback and zener controlled to remove hum and line breathing, the second (if used) in order to remove high frequency glitches, and the third is always open loop and is actually part of the individual stage. It is usually designed to isolate that stage from other signals. It is simple, usually a cap follower, so it MUST have a stable DC input voltage that is supplied by the conventional high feedback zener referenced voltage regulator at the input.
The main factor that seems to be forgotten is that the regulators must remove both very low and very high frequencies . This is difficult with one regulator, at least, I don't know how to do it easily.

It is an active 'pull-down' of an error signal. We sometimes see something like this in motor drive servos, when we want to stop the motor quickly.

The shunt regulator still can be very valuable because it can be much faster responding than the input series regulator, AND it can give a constant current load to the input series supply at all times. Not a bad thing at all.



What a schematic!

I have made a number of preamp designs over the years, but there are problems with DIY implementation. Of course, I published the Levinson JC-2 preamp almost 30 years ago. After that, I designed the JC-80, the Lineage preamp, the Vendetta preamp prototype, and a whole line of Parasound preamps, including the 1100, 1500 and 2000. Now there is the CTC and the Parasound JC-2 preamp, both still in production.
Most of the Parasound stuff is pretty good, but difficult to implement, because each preamp has so many features, and this makes it complicated.
The JC-80 would be a good candidate, but the output fets are not available to anyone, anymore. In fact, most of my best designs use difficult to find or obsolete fet's. It seems that only Erno Borbely and I have any stock of these fets and Erno sells them to audiophiles at a high price. I would too, if I were to sell any.
This poses a bit of a dilemma. If I publish specific circuits, then I will be asked by audiophiles around the world to 'help' them find parts, etc. This creates an almost impossible situation, so I would prefer not to do it. I hope that this explains my reluctance to publish my circuits, especially on a DIY website.

The Vendetta preamp now has C, D, and T mods, since B.
It is now almost impossible to get Toshiba fets in matched pair. I have used k240, K146, k389, and their complements in most of my designs. To specify these now would be a problem.

You have to ask Bear about the teflon silver cable that we use in the CTC. It is VERY expensive.

Input selector

Many of my less expensive designs use relays. Relays are OK, but silver on silver switches are better.

Actually, this is my experience:
Levinson JC-2, Rotary Switches

JC-80, gold on silver hermetically sealed relays
Vendetta preamp, relays
Lineage preamp, Solid state fet switches
Parasound preamps, Toshiba solid state switches
Parasound PLD-2000, relays
CTC, silver on silver switches
Parasound JC-2, relays

I've tried them all, and I think that the scraping action of the silver on silver high tension switches are the best. Your results may vary.
It is amusing, because just 3 weeks ago, at CES, Dave Wilson was going on and on about a silver on silver rotary switch and how good it sounded. I wonder why?

mercury wetted relays - I don't like them, because they have too many dissimilar contacts in series with the relay. They tend to 'forgive' rather than sound harsh. This is, at least, my experience with the so called 'Speigel box'

It is almost impossible to do better than gold contacts in relays at low levels. A great reference is Holm, a German engineer before and during WW2. They didn't name the relay conference, the Holm conference, for nothing

The Shallco switches are self cleaning. That is the point of it all. Relays don't self clean, so you have to use more passive materials, such as gold, directly for the dry contacts. However, the scraping action of the Shallco switches literally creates a fresh surface each time they are switched. Perhaps, once in a while, one should 'exercise' the switches to remove any sulfides that might be taken out of the outside air. Now, our virtually sealed interior in our preamp makes more sense, doesn't it?

I like relays, but I don't think that they are quite as good as those big Shallco switches. I have one of the Shallco switches that we use in front of me, and without a large knob, I can't even make it rotate. That's spring pressure for you.

I LOVE remote control. I have remote controls all over the place, but we didn't want any sonic compromise in this preamp.
The latest Parasound JC-2 preamp designed by Bob, Carl, and me (CTC) has remote control, because many people demand it. We use a cheaper TKD pot with a motor control. It, to us, is as close as we can get, to the CTC preamp and still have all the features that most people rely on.

Servo

Servos can be problematic, IF the effective bandwidth is not low enough. Decades ago, I thought that it would be a good idea to make a servo also act as a hi pass filter. BAD IDEA! It really effected the sound and I found that a good coupling cap was actually better.

I have 2 separate servos in each line amp, because it has differential outputs. One servo tries to keep the average of the two outputs near ground, and the other compares each output and adjusts them to be close to each other. The servos must be very low in frequency cut-off in order not to effect any asymmetrical low frequency transient that might occur on program material. This isn't obvious, unless you look into it, how very low you have to go, in order not to effect things.
Also, servo gain should be the minimum necessary to do the job, but no more. The servo should be considered an 'impure', but necessary, element that should be buffered from the audio signal as much as possible.

It is important where to return the servo to the input.
The further down the circuit path, the better.
The CTC connection is in the second stage, which is really the cascode of the amp. I use 4 separate connections. This tends to make the input to be less effected by the servo, which is probably a good thing. You might call my line stage a hybrid of servo and open loop only.

it should be obvious now, that Jcarr and I, as well as Walt Jung, and probably several others involved in hi end audio, take servos seriously, and don't just add one, without seriously attempting to minimize their potential problems. Some of us don't use servos at all, but I find that impractical for what I am trying to do, ie make noiseless polarity switching at the preamp output, rather than somewhere else in the audio chain.

I agree that the best open loop DC stability is the best way to design preamp stages. My design will work without any servos, but it would have some output offset due to the input matching, which is seldom perfect in fets.

Specifications

Folks, I need at at least 1/2 A at +/-36V. How long do you think that these puny little gel cells will last, both short term and how about long term? I already have some experience in this, so don't just guess.

The specs are not very spectacular. Perhaps .01% distortion at 2V out or so. Output impedance is fixed at 1000 ohms. Input impedance is whatever the volume control pot is.

Transconductance amp

The CTC preamp is a transconductance amp that is current output. 1K ohm was the lowest value that we could use and still get reasonable distortion at a few volts out. There are no followers in this preamp. This was also true in the Levinson JC-2, JC-80, and some other line preamps that I have designed. I do use complementary fet followers in the Parasound JC-2.

The transconductance amp is just like the Audible Illusions open loop triode line amp concept: It is simple.
If you think of a single active stage, that has voltage gain, it will always be a transconductance amplifier. We can make line amps with tubes, like this, why not fets? In order to make an op amp out of this type of design, you have to add some kind of voltage follower, in order to lower the output impedance and to increase the effective gain of the first stage (if possible).
I have always used transconductance amps for my line drives, when practical. They have several other advantages, as well as simplicity.

Transconductance amps are more stable with a difficult load. I made the line driver for the Grateful Dead with this technique. Very difficult load, worked great!. This is because followers tend to ring due to the generation of -R due to phase shift at high frequencies. TA's just slow down, but don't ring very easily.
My designs are also simpler in thru-path than op amps, because they don't have the extra output stage.

Current output amps are standard for DC motor control, but most loudspeakers are designed to be driven by voltage, not current. Emitter resistors are not a big problem, if they are small valued enough.
The biggest problem is output gain. If you have only 2 stages in your amp, then you may not have enough gain to make the amp practical, because the output stage can have negative gain with a low Z load.
If you use negative feedback, then you can develop IIM distortion, where the output from the loudspeaker feeds back to the input and causes overload.

Servo

You should use an attenuator after the op amps, and before the audio stages to buffer any IC or cap non-linearities. I usually use 10:1 to 100:1 attenuation.

Noise

It is true that power amps don't often have much of a noise problem with the input stage, BUT if you used the same topology for a line preamp, or even better yet, a phono preamp, then fet noise becomes very important.
However, this is how it usually works:

Fet noise in Toshiba jfets is usually almost always what it appears in its 'typical noise' graphs. This will be between 0.6 and 2 nV/ rt Hz. at 1 KHz or so. The biggest difference between jfets of a particular type is LOW frequency noise. This can be very different with different batches of parts, but it usually is fairly consistant in one batch, and in general. Sometimes, even Toshiba screws up, and goes out of spec., but not often. Measuring individual fets requires that you run a specific current, and also know exactly what transconductance the fet has at this current. This is not easy for an amateur. It is easiest to use the Toshiba spec sheets. They are accurate. However, some American manufacturers can lie to you, and give you a noisy device, even when the spec sheet looks good. Beware! This is because they changed how they processed the fet over the decades, and never updated (downgraded) their noise specs. I got caught twice with useless parts, because of this.

Substitutes

>One of them was J110, as long as I know it... You pointed them a long time ago with semi-words when trying to help to repair a JC-2

>These parts were Siliconix (the former company)... Then it was bought by Harris Semiconductors... And now it Siliconix again !!!... As an evidence, the production process should have changed with time with such proprietors changes...

>Second part could be a dual high Idss claimed to be a close challenger to the 2SK147... Was a Crystalonics C413 if I recal fine, a part that was later new labelled in 2N6550...

That's pretty good research! Actually, Toshiba made a bad batch of 2sk147's about 15 years ago. I think that they cleaned up their act, but it was a problem for awhile.

I parallel large fets to get lower noise. A Vendetta has 4 large Toshiba fets on the input. I have a noise analyser, so I can pick out the lowest noise units and put them at the input. Signal is correlated, noise is almost always uncorrelated, so it tends to 'interfere' with itself and not increase as fast as the signal does, when you parallel fets, (or bipolar transistors).

All discrete, except for minidip IC's. Teflon board double sided, heavy ground plained, 6 to220 case mosfets on bottom of line stages. All on same side of Vendetta gold plated boards on left.

Noise on BJT

I discovered that 2n4401-3 transistors were extra quiet , back in 1968, and I used these parts in the Levinson JC-1 pre-preamp in 1973. However, I did get a footnote in about them in 'Wireless World' in 1971-72 sometime, connected to another article on noise. They are good parts, but Rohm and several other parts are better. It all has to do with base resistivity, or Rbb'.

I first discovered the low noise properties of the 2n4401-3 transistors when working on low noise electronics at Ampex in 1968, soon after these devices were released by Motorola. I had a Quantech noise tester available, and I tested everything that was released that looked promising. It wasn't a well known low noise part, until M&F published their great book on low noise design in 1973. It was used in the Levinson JC-1 pre-preamp, that was first released in 1973. For years this was the best part-pair, but in the late 70's Hitachi made a really super part-pair, that is still the best ever done, as far as I can tell. However they discontinued the parts, decades ago.

The best 'modern' paper on semiconductor noise is: 'The Design of Low-Noise Amplifiers' By Yishay Netzer, 'Proceedings of the IEEE' Vol. 69, NO 6, June 1981. This is a good example of: 1. It doesn't exist. 2. It exists, but it is not important. 3. We invented it, or in this case, it is obvious.

I might point out a few things about low noise transistors.
First, transistors have 3 significant sources that must be minimized, if possible. These are: voltage noise, current noise, and base resistance.
Many 'low noise' transistors have high beta. This is important to reduce current noise, which is proportional to base current. However, high beta usually also means high base resistivity, so these devices are always more noisy with low source resistances, such as moving coil cartridges.
The 4401-3 is a low-medium beta device, so it is almost worthless for line amps, power amps, etc, BUT it has one major advantage: It has low base resistance of about 40 ohms, rather than 400 ohms, typical of early low noise devices.
Today, there are many devices, usually Japanese, that are pretty good general purpose parts. I don't have specific numbers, because I almost always use fets, as they are just as quiet as a 4401-3, at every input impedance.
Please remember, the 4401-3 was just about all there was for low inpedance design, for about 10 years, maybe from 1968-1978. Then Japan made better parts, Hitachi made a super part, but they never got released openly, so it would probably be a waste of time for any of you to try to find them. Rohm has made many good low noise parts, but some of these are hard to get, as well. For serious low noise design, it is best to use fets, instead, or maybe a Rohm part, if you can find them.

Active voltage regulators

Typical active regulators have many problems. However, they are cheap and can give a reasonably accurate output voltage.
These problems include:
They are noisy, because the reference is unbypassed.
They respond to changes in current with transient overshoot.
They are virtually out of the circuit at low RF or CD clock frequencies.
They can easily pass fast line transients.
This is OK?

It probably depends on what type of circuit you are using as a preamp. Some are more sensitive to power supply than others. I KNOW that my Vendetta input stage is very sensitive to power supply regulators. I have done A-B comparisons. For a simple IC based line amp, the power supply is probably less important.

Quality discrete solid state design is getting more difficult each year. This is because quality parts are becoming more difficult to buy. I don't really try to hide my circuit designs anymore, but I just don't publish my latest schematics.
One reason is that many people, when they try to copy something, will blame the designer, if they get into trouble.

My circuits are simple, but they require selection and matching of complementary fets. It is difficult to get a successful match, unless you have 100's of fets to chose from. This is not only expensive, but it is almost impossible to get this many parts from scratch.
My Parasound designs are made with similar parts, but the circuits do not require the matching or the Idss range, that my CTC design demands. This is why they can be made in Taiwan by the 100's.
I have found that the most simple designs, that are also optimized for lowest distortion, are best, sonically. This includes triodes, as well as solid state, especially fets. Fets are significantly less linear than triodes, so I have to make complementary circuit paths that help each other, to obtain the same distortion as a good open loop triode. The only real advantage that I have is that I can direct couple both in and out, without caps. Also, my fets don't seem to age (get tired) over the years.

Capacitors

For the record, I don't like to use caps in the audio path. They are too darn expensive, and they still give problems. I can direct couple, so I do. That's called 'elegant' design.

Cascode

I usually run fets around 15V, or about 1/2 their voltage rating. I do this because they become more linear at high frequencies with MORE voltage, but then they will start to leak gate current, if too much voltage is applied. Look at the detailed data sheets of specific fets and the change of Cgs and Cdg with voltage. You will find the 'rate of change' to level off after several volts. Very low voltage operation gives you high value, non-linear capacitance from inside the fets.

A self referenced cascode is a good compromise design component. This is where the cascode is made by a fet with its gate tied back to the input fet's source.
Usually, we use 2 different fet types for this. A short gate, low Vg part (hi transconductance) for the input part and a long gate (low transconductance) part for the cascode part. Identical parts may be an interesting solution, but I tend to avoid it. The input capacitance will be very high, BUT the change in drain-gate voltage will be very small, so the non-linear capacitance will be essentially invisible. This is not true with a non cascode fet that has to swing significant voltage at its output. Here, operating voltage across the fet is very important. Of course the best is to operate the fet with a reasonable voltage, and then cascode it as well. Best of both approaches.

Servos

You should not create an artificial ground and then run audio through it. It will map itself into the audio signal itself, i.e. any impurities generated by the servo IC will be part of the audio path. No wonder you can hear these servos so easily. Always separate the servo from the audio thru-path. This can be done by adding a low value resistor to ground, and the servo resistor high valued. Then a small imperfection in the servo will be attenuated many times before it is added to the audio path.

It is important to ACTUALLY attenuate the servo output so that the servo must work many times more in voltage for the correction it is trying to implement. For example, a 20mV input offset might take 2V of swing on the servo, and the servo will sit at 2V out most of the time. This way, the servo is not adding itself to the input signal in any significant way. This is because the servo IC's and caps are usually inferior to the actual circuit in their transfer function, so you don't desire that the IC servo's sound is added to the audio passing through.
I completely understand that some of you, like Elso, have had bad experiences with servos. I would not use servos either, if they were not necessary in my designs. Please remember that I make components for others, who are sometimes very sensitive to a click or pop to the point of neurosis. I just can't afford to allow my absolute polarity switch to click or pop just because I felt that 10 mV of offset was OK.

337

In principle, the 337 as a current source could be OK in many applications. However, consider this: This active current source will be noisy, spit transients, and probably not be a current source as you approach 1MHz or so. This is because this current source is created by an op amp that is made of the cheapest parts possible.

Open loop preamp

I would like to make an all 'open loop phono preamp' but it is not easy to do right. Someday I hope to make an improved replacement for the Vendetta phono stage. Now I am just updating units.
I find the constant criticism of this preamp, virtually a joke. We are now building 5 of these units, and I can say that the case is a little larger than necessary for most versions, but my personal unit, with the Vendetta phono stage is fairly full up, actually, and it would be even more so, if I had selected to have balanced inputs as well as balanced outputs. These units do not generate any signficant hum pickup in actual operation.

Balanced inputs

Balanced inputs are overrated. Actually the electronics circuit is fully balanced on both input and output, but virtually all of my sources are single ended, and so it is better to have single ended inputs on my preamp. However, the same CTC team has designed a less expensive preamp for Parasound that has fully balanced input and output. This requires XLR connectors on input and output, as well as quad input pots and selector switches. Think of the extra cost and space needed for a preamp of the Blowtorch's quality. It just wasn't worth it in my case. Now what about hum? Well, there isn't any signficant hum, but then I don't put my power supply next to the preamp either. I put it on a separate shelf perhaps a meter or so away from the preamp. This is necessary, because it is the hum sources that must be avoided.

It is true that with an all balanced system, the power supply could be moved closer. This is a small improvement for me, but your needs may be different.

IC regulators

I tend to not complicate designs more than what I think is necessary. I don't know if the more complicated regulator would really help, but I would doubt it, in my case.
This is because I am making fixed regulators that are only optimized to filter input hum down to the noise floor. I know that these IC regulators 'fall apart' over approximately 1KHz and get worse and worse as regulators. They also have marginal transient response, both input and load. They will also ring as they generate a synthetic inductor at their output. A very high Q cap can make things actually worsen at the output of these regulators.
However, let's say that I wanted to make a 'lab' supply: Then the situation might be different. I might try to remove any hum or noise to the highest degree possible so that I could test a prototype preamp design without any extra regulators downstream.
Personally, I find the IC regulators marginal and I like to separate them from the actual working circuits. However, they DO reduce the hum and set a respectably constant reference voltage. To reduce transients, RFI, and audio noise, I use other circuits or passive parts either before or after the IC regulator.

Zeners and FET current sources

However, let me point out where some things could go wrong:
First, zeners are generally very noisy devices. Where 4nV/rt Hz might be considered a 1K noise source, a 15V zener diode might have 2000nV/rt Hz noise. However, a 3V zener is pretty darn quiet, because it relies on the actual 'Zener effect' rather than an avalanche effect. Over 5V or so, zener diodes get really noisy! Of course, with this current source, you used only 3V, so you are in good shape. That is why I had to ask for specific values, in order to see if there was any problem.
It is also important to see this circuit as a potential amplifier of its own self noise. Without the 10 ohm resistor in the emitter, you would have a noisy current source. This is a frequent oversight made by many designers of active loads, etc. However, with the 10 ohm resistor you have dropped your potential noise contribution by the current source about 10/.125 or about 80 times. That 10 ohm resistor pads the transconductance of the transistor that much! Perhaps, you knew that, Richard, but I hope that I can point this out to others, who might be looking at these circuits.
Finally, I would have used a jfet as a constant current source instead of the 1 K resistor. The actual current is not too important , 2-10ma would probably do, so long as you have enough voltage across the fets to keep it acting like a current source.

It is possible that the 220uf cap on the current source could throw things off. I think that a J203 fet could replace the 1K resistor. It is true that 2ma is a bit low, but with too much current would give too high a Vp and make for a higher voltage initial supply. It seems to me that the shunt fet is really an open loop follower, so the output impedance of the regulator is really 1/gm of the fet itself. The 220uf cap in that circuit is probably OK.

I know how to make cheap stuff as well as expensive designs. If I am to make an all-out design, then I will go all-out on the regulators as well. When I help Parasound with a cheap-to-make design, I am much more forgiving.
I am trying to make power supplies that will effectively be as good as a quality battery. Many here wonder why I don't use batteries, then. Well, batteries are unreliable, and I know this from experience. Also, I like to run my designs with higher current than many other designs, when I can. This is partially because the parts that I can get are really designed for higher current operation, and I like to operate closer to their optimum operating point, and I usually find that the distortion drops both in amount and in harmonic order, when I run at higher currents. This makes batteries impractical, except for really crazy people, (like some of my friends). Now, what can't I do as well as batteries? Usually, it is isolation from the power line, which I really think is pretty dirty. I have measured my own power lines, and found them to have garbage in relatively high amounts, not only at low freqencies, but at mid frequencies 5K and above, as well as RF. Why do any less?

It is called: 'Rectifier and Zener Diodes Data' 1984. It seems that the very first quality zener diodes had a noise curve measured on them. So, if you took a particular zener diode series, you could estimate the noise contribution.
One series is the 1N746-759 and 1N957-1N986.
On page 4-8 they show a 'typical noise density' graph.
Below about 4V, the noise is so low, that it is off the graph. However, the optimum operating current of these devices is about 20ma. 2 ma operation will severely effect the operating impedance, changing it from 10ohms to perhaps over 100 ohms. Not too good. Above about 4.5V the graph turns upward fast! At perhaps 1uV/rt Hz at 4.7V, to 1000uV/rt Hz at 10V! It peaks at about 15V to about 2500uV/rt Hz, and then DROPS to about 50uV/rt Hz at 40V or so, then starts up again and peaks at 1500uv/rt Hz at 90V. What a curve!. Is this typical for today's devices? I wouldn't chance it, and ignore this curve.
In any case, for low noise, stay below 5V or so. I should think that leds would actually be a better choice. This is where #75 could be valuable. Also, any input by Walt Jung would be useful. However, wouldn't noise add as the sqrt of the sums of the squares, rather than linearly? So, 2 identical diodes in series would have 1.4 times the noise.
I only glanced at #75, so this may be the same thing. I'll go back an look.

Long gate fets, like the j203 or the k246 are better current sources, because they have LESS transconductance and have generally higher output impedance. However, they are limited in current. I would think that an led or two in series, would be best.
Zeners are very susceptable to operating current, not because of noise, , but because of impedance, which reduces their effectiveness in regulating anything. Noise in zeners is facinating, because it varies so much with operating voltage.
Looking through my book further, I found that the 1N4099 thru 1N4135 series of 1/4W zener diodes gave a slightly different spec.

For example: 4.3V gave 1uV/rt Hz, 4.7V gave 1uV/rt Hz, 5.1V gave 2uV/rt Hz, 6.2V gave 5uV/rt Hz, and 6.8V and above, gave 40uV/rt Hz! This is better than common zeners, but the trend is the same: Use zeners below 5V, or else you must expect noise. Again, I would stick to leds, but there could be a 3V zener out there that is actually quieter than 2 diodes, but I expect you would have to run it at a higher current than the leds to get the same performance.

what FETs you are talking about

I use the 2sk146, j73; 2sk147, j72; 2sk170, j74; 2sk240, j75; 2sk389, j109, and many more. I prefer V for my best designs.

Noise in Zener

For the record, reversed biased transistors have been used as 'zener' diodes since the first analog IC op amps were designed. It is a convenience, because additional processing steps do not have to be done, in order to create a 'zener' reference. This goes back about 40 years.
Now what we have established here, as far as I can tell, is that leds are probably the best way to reference a current source, unless you have a current limitiation, and this is where some special low voltage zeners, operating at 250 ua seem to be OK, except they will be fairly high impedance at their operating point, and therefore useless in our present current source examples.
I wish to point out that zeners are invariably rated at micovolts/ rt Hz, especially above 5V. To convert to rt/Hz, just divide Christer's measurements by 144, (I am guessing here, but that is the sq rt of 20KHz) Even 1uV/rt Hz is REALLY noisy. Most IC op amps work 100-1000 times quieter than this. My original Levinson JC-1, Sota headamp, and the Vendetta phono amp are rated at 0.4nV/rt Hz. This is 2500 times lower than a good zener. This is why I used a Norton equivalent of a zener instead. There I have a low Gm, low noise fet, like a J203, as the reference, with its noise attenuated by a large (100uf) cap to ground. It is true that there will still be some really low 1/f noise, but it works well in the audio range.

TL431 noise

I would put a 1K resistor just after the TL431 output, then the cap. The problem here is: The TL431 makes 100+ nV/rt Hz noise and has a very low output impedance. Putting a cap DIRECTLY across the output will NOT do very much, if anything useful. The cap must have something to work against to be a good RC filter. Now, 1K and 220 uF for example, gives a rolloff starting just below 1 Hz, so at 100 Hz, you have about 20 dB lower noise from the source, and at 1KHz, you have about 40dB or 100 times lower noise. For a line amp, this would be OK. A moving coil pre-preamp should use a completely different design and reference, because the bass frequencies can be boosted as much as 80 dB, or 10,000 times. In this case, the 1/f noise that would remain, would be annoying during measurement and even listening.

This oversight is trying to bypass a voltage source with just a capacitor. It just doesn't work very well at audio frequencies. Yet, people, both pro and amateur, will try to bypass a zener, or a TL431, or LED's without thinking it through. Of course, it is best to start with a low noise source, if it is practical. When it is impractical, or all that you can get is some component, then knowing how to apply an effective low pass filter is useful. We can go even further with active cap multipliers, or even an active low pass filter, but that is a lot more work than a simple RC time constant added after the voltage source.

FETS

I feel that it is easier to use a mosfet for the output devices. This is because they have inherently higher voltage breakdown, and can be found in some sort of 'power' package. J-fets could be made this way, but they are not normally available, except in smaller packages.
I have used the Hitachi 213-16, J76-79, for many years. The P channel devices, unfortunately, are more like triodes than normal fets. This will add a lot of second harmonic distortion to a typical balanced circuit. Still, they are pretty good, if you watch out for this.
In general, I have found it better to make simple, balanced circuits, rather than complex circuits with multiple junctions in the thru-path. That is why I use such a simple circuit, rather than add followers, etc. It depends on what you are trying to do. If you need high voltage gain, then adding a follower can give you all the gain that you usually need. However, if you only need a gain of 10 or less, open loop, then adding a follower stage will be redundant, unless you need really low output impedance or to drive very long cables.
While I used a design technique that I originally found in Charles Hansen's design, I am really trying to parallel (in solid state) the successful tube output stage, first set forth by Audible Illusions. I worked with Audible Illusions for a few years, part time, and fell in love with their open loop tube output stage. I tested it more fully, as well as listened to it at home, and decided that it was a 'winner'. My contribution to the 'Blowtorch' preamp was to make a fet version that could behave well open loop, with a minimum of distortion. Instead, of a single tube, I had use a complementary differential folded cascode. Quite a trade-off, but I did get to eliminate the output coupling caps, and get balanced output drive. The measured results are similar.

I am not saying that the Hitachi devices would not work in this case, but to understand that they are prone to output impedance mismatch, in general, and this could be a problem in some designs. Some of the Toshiba 1A parts look very interesting. 2SK2013 / 2SJ313

All solid state devices have a finite output impedance when biased in the linear region. We often compare the different output impedances to triodes to pentodes, or somewhere in between. If you look at Hitacihi lateral 1/2A mosfets on a semiconductor curve tracer, you find that the output curve of the Hitachi P channel looks like a triode, and the Hitachi N channel looks like a pentode. If both were 'triodes', then there would be a better match, alternatively, if both were 'pentodes' there would be a better match.

Since the output impedances are not matched, it is difficult to use these devices directly, in push-pull, as gain stages. However, they will work fairly well as a cascode, since the output impedance will be much increased, just because the device is operating as common gate, rather than common source. Then the mismatch may still be measurable, but within reasonable limits to be useful.

If you are using a folded cascode design, then you are already more than 1/2 way there. To cascode a cascode may cause more problems than it is worth, in most cases.

ost of the distortion should be from the change of Gm with current. The lower the output impedance, the more the current will change in the input differential pair for any given output voltage. This means more distortion with lower impedance, and why I stopped at 1K.
Nonlinear capacitance is real, but minimized in this example. This is because of the cascode, AND because there is significant voltage across the input fets. Now you know why I run them so hot, close to their limit. This is NOT how I design high production designs for Parasound, but I have never had a problem to this in limited production.

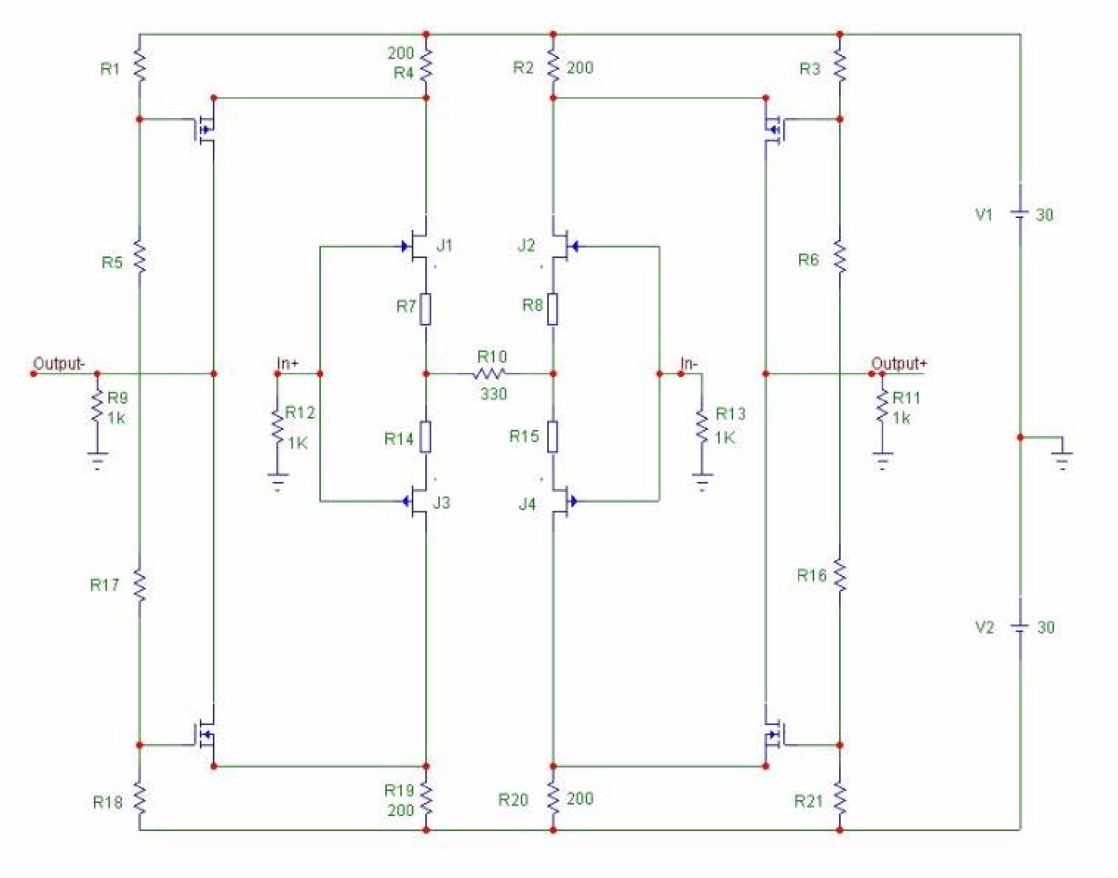
It is somewhat better to actually measure a pair of devices, but they would measure fairly well anyway. It is more important to understand the type and order of the distortion that is generated and whether it would be audible in any case. Many times, people worry about very small numbers, but that is not what makes the difference in sound quality. A small amount of lower order 2nd or 3rd harmonic distortion is very difficult to hear.

I try to use +/- 30V supplies. This drops in half when actually operating each device, because the 200 ohm drop resistors need so much voltage to get the current necessary.

I chose mosfets for higher power dissipation and higher voltage breakdown than possible with jfets.

TO-220 sized Hitachi lateral MOSFETs.

It is actually more like 10V across each 200 ohm resistor. I forgot to subtract the input-output voltage differential of the final fet follower. Yes, more current is better than less current in the output devices.



This is a simplified version of the line amp that we used in the CTC preamp. Does anyone have a computer emulation of this circuit? I don't, for a number of reasons, but I know what it measures. It would be interesting if the emulation looked almost exactly like the measured output.

Actually, the curves for the j76 do not look OK. They should be more 'triode-like'.
As far as exact values are concerned, I am not giving them out. Not because they are so special, but because I specified, up front, that I was not going to publish the specific schematic of this line amp, as long as I am still producing them. The simplified schematics presented are a good base to discuss the circuit, and even for someone to make something that should work well on an individual basis.

The biggest 'secret' is 'elegant simplicity'. Try it folks, tends to sound better than complex circuits.

Folks, over the years, I have found that' less is best' for audio circuits. Now, if I were to make a low distortion amplifier for test signals, then I would do a more complex circuit, with lots of feedback. I've done it for a major test equipment manufacturer. How about an open loop bandwidth of 100,000Hz and virtually no distortion (less than .001% at 100KHz at several volts out?

I recall that we discussed both power supplies and servos to a great extent on this thread. It is true that this type of circuit is more sensitive to power supply variations than some others, but it is the most linear way that I know to make a fet gain stage without global or other loop feedback. That is why I used it in this preamp, but of course, it means more time and effort into the power supply.
This is not a practical circuit for mass production.
Now when it comes to why the preamp itself costs so much, it is not because of the cost of the line circuit chosen. It is because the case, pots, switches, and assembly time costs so much.

I don't understand why caps need high current to drive them, UNLESS they are connected to ground on one side. I like high current, in order to drive long cables with their relatively high capacitance that goes to ground. Coupling caps don't have this problem.

What I am trying to point out is that coupling caps only have a dV/dt problem with very low impedance loads. If you are coupling from a source to a load, either directly or through a cap, it makes little difference, current-wise. It only makes a difference IF the load is too low or actually ground. Then, direct coupling will work as badly as any cap, or even worse, you will be driving a short-circuit. In either case, the cap does not REQUIRE more current. Caps have other, REAL problems, and, by the way, I like high current too, just to drive the residual cable capacitances with ease.
However, a related situation is VERY important. This is the capacitance to ground of long cables, or the RIAA caps in a traditional phono stage. For example, the DYNA PAS-3 phono stage is SLEW-RATE limited by the RIAA cap(s) attempting to drive a 1K input resistor from a 100K load resistor. What a problem! I lived with the DYNA preamp for 10 years, before I realized this. No wonder it tended to 'soften' the sound.

The important thing is to make the most linear circuitry possible, so that feedback is not very important or even can be reduced to almost nothing. This seems to make audio circuits sound better. This is true with tubes, transistors, or fets.
In general, tubes are the most linear individual active devices, followed by fets, and then bipolar transistors. The problem with tube circuits is that they have to be coupled between stages with caps or transformers. Solid state circuits can be direct coupled, but they are always prone to more nonlinearity, both static (from Gm variation) and dynamic (from nonlinear capacitance with voltage). This is where solid state circuits can be optimized more than is typically employed, by consideration of the non-linear capacitance and its minimization.

Negative feedback causes a number of secondary problems, such as: TIM, IIM, FIM distortions, that the ear appears able to easily detect. Please understand: I have designed with negative feedback for the last 40 years. It is just that I have found that, all else being equal, less negative feedback is better sounding than more negative feedback.

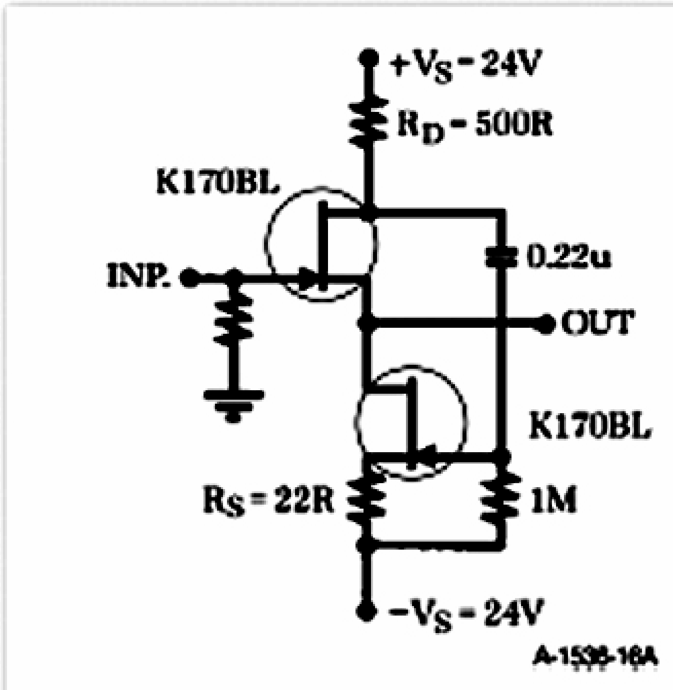
Folks, I apologize, but I can't publish my schematics. I don't mind discussing general aspects of this design, especially in its simplified form. I also don't mind constructive

suggestions as to how to improve it, or change it in some other way. Many of you on this thread, have done a good deal of 'reverse engineering' from the photos, and some good engineering development on how to make a design like this. For this, I commend you, but I can't give you EVERYTHING!
We are still making this preamp. We just finished 6 units this month, and we may build more in future, so my new business partner gets VERY angry with me, if I spill too much on this thread. We are also going to make a new phono stage, one of these days, so I don't want to be very specific about the Vendetta Research phono stage either.
However, I am interested in answering specific questions about audio design, and learning as much from you, as you might learn from me.
Now I thank Poobah for sticking up for me in this case. I might remind him, good naturedly, that criticism of our wiring practices, first caused me to comment on this thread. Having designed numerous preamps over the years, shows me that we are still on the right track. Please note that our wires are NOT shielded. This makes for a better sounding wire, in our estimation. Also, we MAXIMIZE the wire spacing, and minimize the circuit board traces. Circuit board material, even teflon, is not as good an insulator as pure air, and we can keep the spacing between our wires (in air) at a much greater distance from each other. The thick almost airtight aluminum case is our electrostatic shield from the outside world. Does it look 'pretty'? Not really, it is darn hard to make it so. I am in charge of making it 'presentable' so I know this for sure. However, it works and is one way to do things. We still make designs that eliminate wiring altogether, if we can, but they are more potenially compromised, because of this. The JC-1 power amp and the upcoming JC-2 preamp (Parasound) are examples of 'wireless' layout, by the very same person who designed the Blowtorch circuit boards. You just can't have very much individual wiring in mass production, and get it right. At least, that is my experience.
In any case, let's talk about audio preamps in general, not just about my specific design.

No, there is not an exact schematic of the 'Blowtorch' on this thread or any other.
As I stated before, I generally don't publish my schematics, because I don't want to be obligated to bail someone out, who tries to make one and gets into trouble.
I previously stated that this is not an easy design to make. One of the main problems is the 'cherry picking' or limited availability of the fets used in the design. I can only use the highest Idss devices in this design, because the output follows the input, and this is what gives me some current drive capability. I can use only about 20% or so of the V series parts that I have on hand, and then match them to each other. The rest must be put aside for another project. Think about what that would cost most of you to do the same thing. I can just hear the whining now! So, either you would make a compromised version, and then be disappointed with the results, or you would demand an alternate solution from me. I would rather avoid the responsibility entirely.

hope that we can stay on track and discuss audio topics that make exceptional designs. After all, if you are going to DIY something, then you should make the best audio design that is practical for you to make. It is cheaper and easier to buy, new or used, typical audio products at the local hi fi outlet. Why waste your time with making anything less than the best than you can make?

4.5 Borbely follower



Borbeley did good research on followers. His "White follower" comes from a MIT publication written on tube circuits developed in WW2. The White follower uses a .22uf (or so) cap to turn the normal current source load into an active common source amplifier. This gives push pull operation and improved current drive, because the current source tends to turn off when the normal follower fet is driving the load with current.
In my opinion, this overcomplicates the usual requirements for a follower, and I prefer a follower without the added .22uf cap. However, in some cases, this could be a useful addition.
For most amateurs, and me as well, the simple 2SK389 slightly biased off from Idss by two 10 ohm resistors, running at +/- 18V, works darn well with loads over 5K. Higher supply voltages could cause problems with leakage or breakdown, yet could be used with selected devices, but not by amateurs without a curve tracer to select out high voltage parts

We use silver solder to improve the contact between components. I know this is beyond you at this time, but it is will known in hi end audio. I use it, myself.

It is best to use the White follower or even the simpler all N channel follower also described by Borbely. The same substrate is best. Just use a 389. The complementary follower will give you significant DC offset.
If you can, you should use SN 62 silver solder, which is what the real pros use. It melts at lower temperature than normal solder and will not make bad solder joints. It also sounds better.

It is almost impossible to completely match complementary fets. You can match for Idss, of course, but any mismatch will cause offset. Also, because geometrically identical complementary fets have different G(m) as the mobility of electrons of N material is significantly different than holes in p material.
Matched N channel fets track with temperature and cancel their effective offset, if identical resistors are in their sources. This is because their G(m) is identical as well.
For the record, forward biasing is sort of OK with many modern devices, such as the 2SK389. This is because of the relatively high G(m) at Idss. In earlier years, this would have been frowned upon, because with the relatively low G(m) akin to the 12AX7, the input impedance of the fet would have then been very variable and leaky. However, high power RF tubes are driven in this condition, even today.

If you want power transmission, then the complementary follower is generally necessary, but the White follower would do a pretty good job. The White follower IS push-pull, not single ended. This is why it can drive lower impedance loads.
My standard follower, composed of a 389 and two 10 ohm resistors works with loads of 2K and above, fairly well. It falls apart at lower impedances.
If you want to build a headphone amp, then paralleled 2sk389's and 2sk109s could work OK. V's (high current) are best for this application.

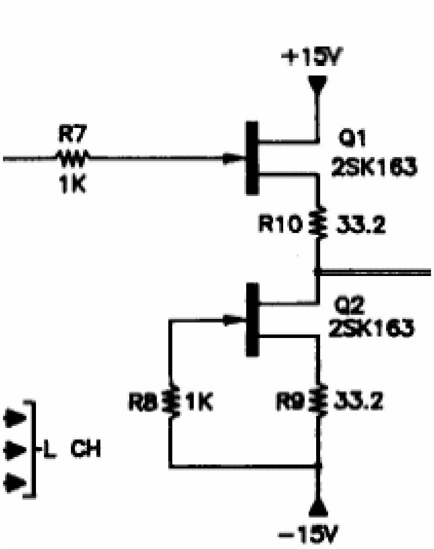
First: 'White' is a man's name. Hence, the 'White follower'
This TOPOLOGY was designed before fets were available, back in the 1940's or maybe earlier.
MY TOPLOGY, as described by Borbely, is a fully complementary fet pair with the gates attached together and the sources attached together.
The TOPOLOGY that I recommend is a VERY SIMPLE fet follower with an active FET load. It is best done with 2 low value resistors (10ohms or so) and a MATCHED FET pair. This works for most applications, with a load over 2K.
This approach is SIMPLE, CHEAP and has VERY LOW DC OFFSET.
If you folks want to actually learn anything about circuit design, please just try this simple follower design. If it is not adequate, then try the other designs. After all the 'White follower' is very similar to my simple example, except for 2 added resistors and a .22uF cap. So try that next. It will drive a lower impedance load, BECAUSE it converts the current source load fet into an active drive fet and you get 'push-pull' operation. The inherent weakness of this design is the added AC coupling with the added .22uF cap. The initial simple circuit can be DIRECT COUPLED on both input and output.
The complementary fet follower can be very linear, but usually will give some significant DC OFFSET on the output, with normal set-up and operation. It could be SERVOED to remove the DC offset, but that is beyond the scope of this discussion.

It is the offset of the original fet pair.
You can adjust it out by SLIGHTLY changing the resistance of the bottom (- connected) resistor. Still, this is darn good!

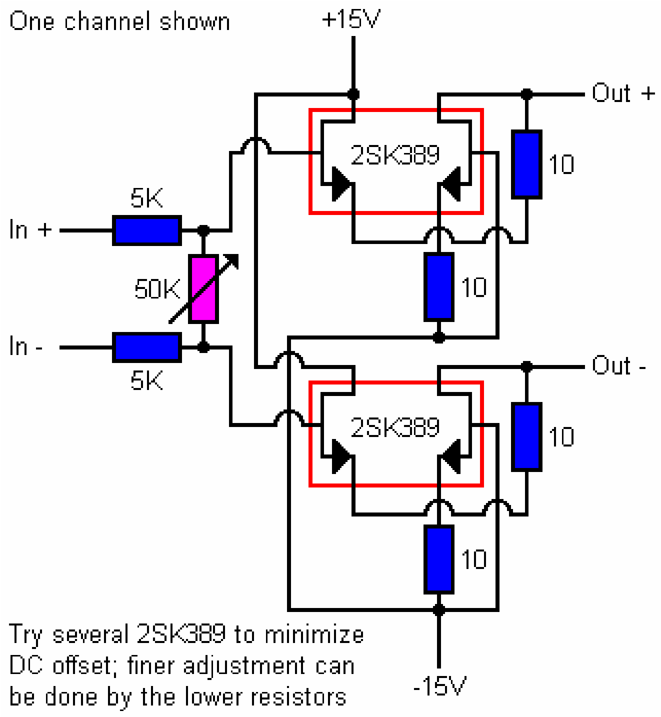
Of course I use a bipolar supply most of the time.

We were designing a FET FOLLOWER. Now, what is the best approach? Erno Borbely has written some of the best input on this subject, however the 'White follower' diagram submitted by 'Frank' is truly classic, in my opinion. Folks, please look at that diagram carefully. It is very sophisticated.
My follower suggestion, which runs counter to what I am known for by Erno Borbely, I have found to be almost perfect for most applications. It is simple, direct coupled, low distortion with a relatively high impedance load, quiet, small in surface area, and self limiting, so that it doesn't break easily.
'Complex simulations' and 'second guessing' only complicate understanding what to do in this simple situation.
My most personal suggestion is that questions should be thought through a little more before addressing your posting to me. I don't mind questions, but I would appreciate that I don't have to work up an obvious answer on a regular basis, if it could have been initially avoided if the questioner would have looked at the problem a little more carefully. This has nothing to do with answering regular questions, I am happy to help.

distortion can increase somewhat [with load], but it will still be small and low order. Actually a follower 'bootstraps' its own gate-source capacitance, and doesn't have 'Miller' distortion, so it can still be darn linear, even with increased input resistance from the attenuator. Remember, one side of the attenuator is always at ground and the other side is always connected to low impedance source, so the actual impedance seen by the follower input is the parallel combination of the 2 resistances. So a 20K resistor and a 100K pot would have a MAXIMUM resistance of only about 30K. Not really any higher. This is reasonable for a follower. NOW, if you wanted to make an attenuator with 10 times higher value resistors, then you might get into trouble. Don't worry, be happy! ;-)



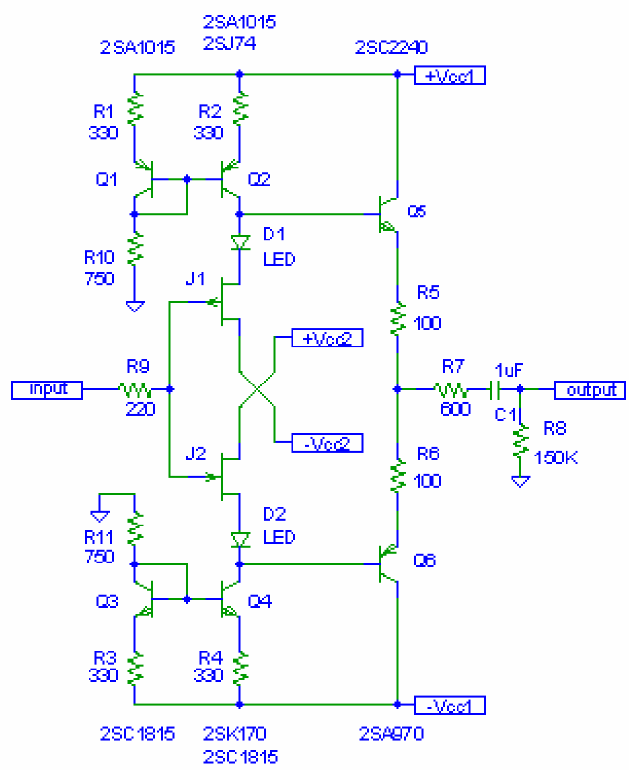
I think that the extra 1K resistor is a useless oversight. All it adds is extra noise. In some exotic application, or with mosfets, it could be useful, but 1K is still a pretty high value. The 1K resistor adds 4 nV/rt Hz noise to the input of the fet current source. This seems to be a carry-over from transistor circuits, where the significant base current could create an offset, if the two active devices are not equally loaded. With fets, this is not important.

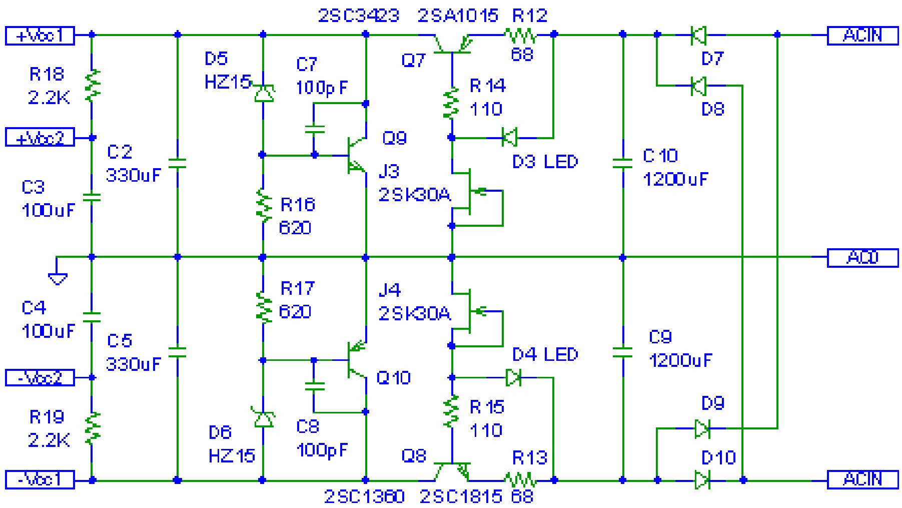


It will only work with a balanced signal, but that's OK as well.
This simple fet follower has many qualities, and some drawbacks. The positive qualities are: simplicity, class A operation, bias stability, and very low DC offset (with matched parts). The limitation is low tolerance for low resistance loads, below 2K or so. My associate, Bob Crump, successfully used this follower in a preamp for several years.
The next level is the complementary fet follower. This has the advantage of better current drive, and cancellation of even order distortion. It has a problem of being more difficult to control offset. It is also not as self limiting as the simple fet follower, but it will usually self limit, before it breaks.

4.6 Various design opinions

Follower

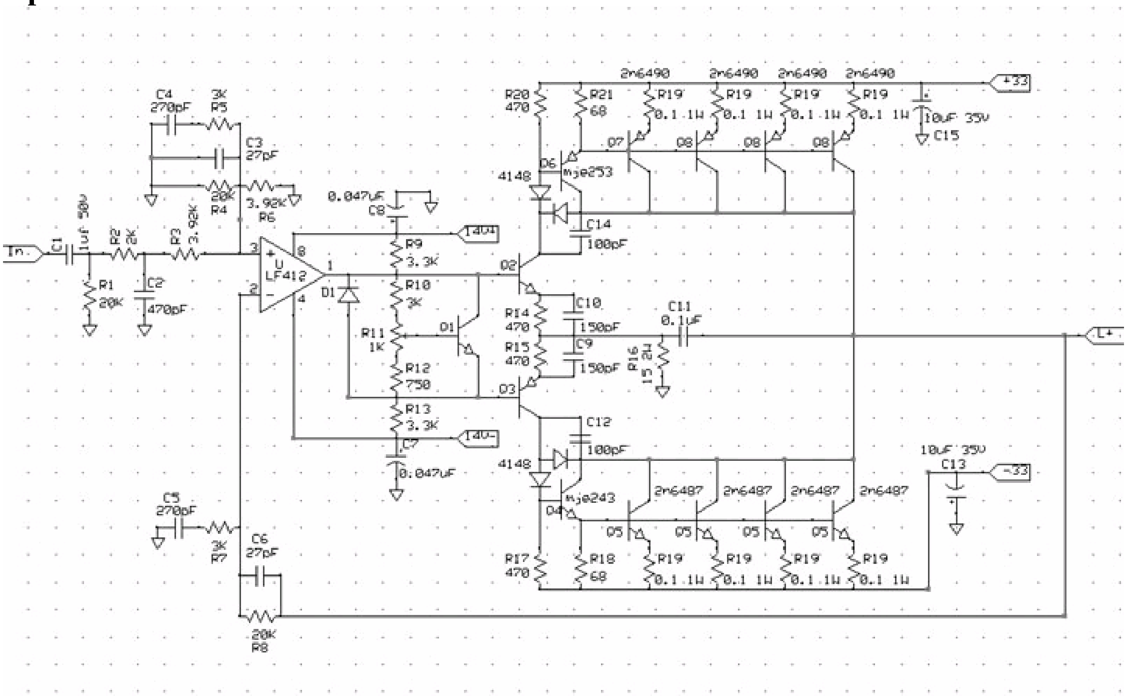




Personally, I think that the ORIGINAL follower design is fundamentally wrong, because it does not naturally follow the characteristics of the devices used. This does not make it a bad design, or that it might even have some qualities that we might find important. Still, a comp j-fet driving a comp transistor output in a Darlington type configuration would work as well, and use less that 1/2 the parts.
The type of circuit originally posted works really well with all bipolar transistors. After all, it removes the need to add bias diodes, etc, to the input. When you use j-fets instead, you create a need to bias the j-fet which means adding led's. This is unnecessary, if you

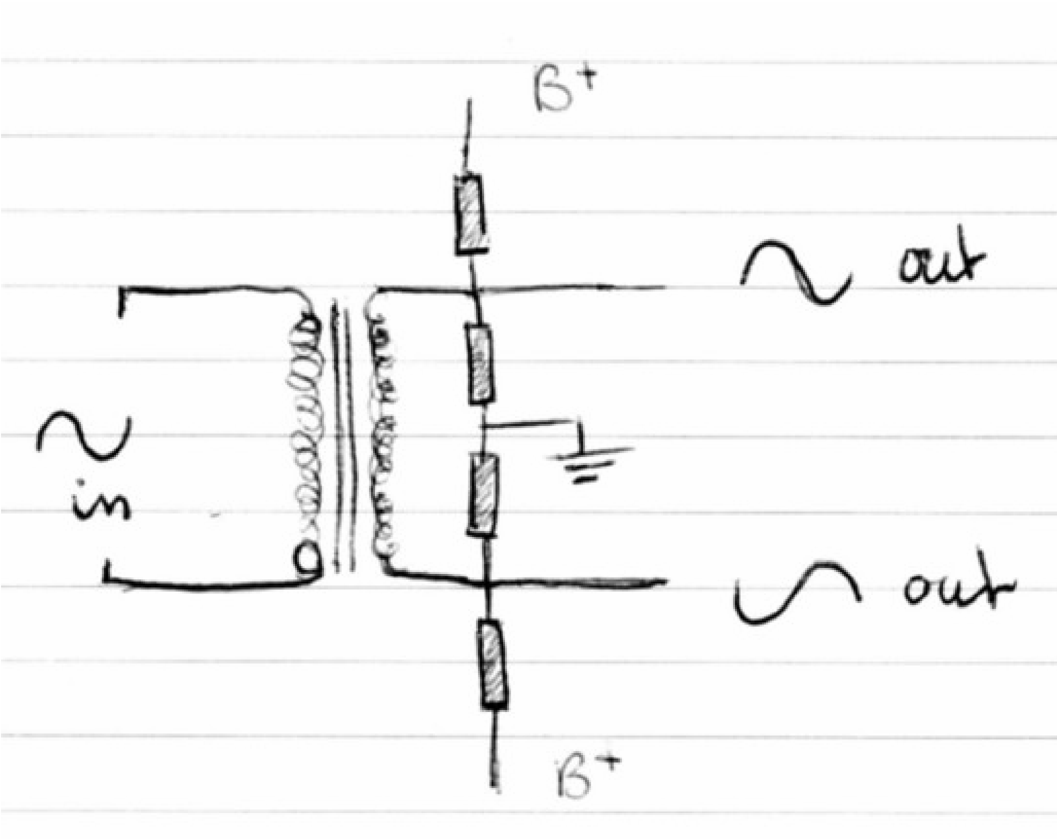
just invert the polarity of the fets and make a comp j-fet input follower driving a transistor pair. It can be done. Then the current sources are unnecessary, led's, etc. For the record, many complementary j-fets have been made to match each other, especially with Gm. The trade-off is extra capacitance on the p-channel device. I hope that this helps.

Opinion



This is a competent design, but perhaps not the best for audio. It would have a high measured IIM distortion. Of course, the 412 can be updated to a newer part.

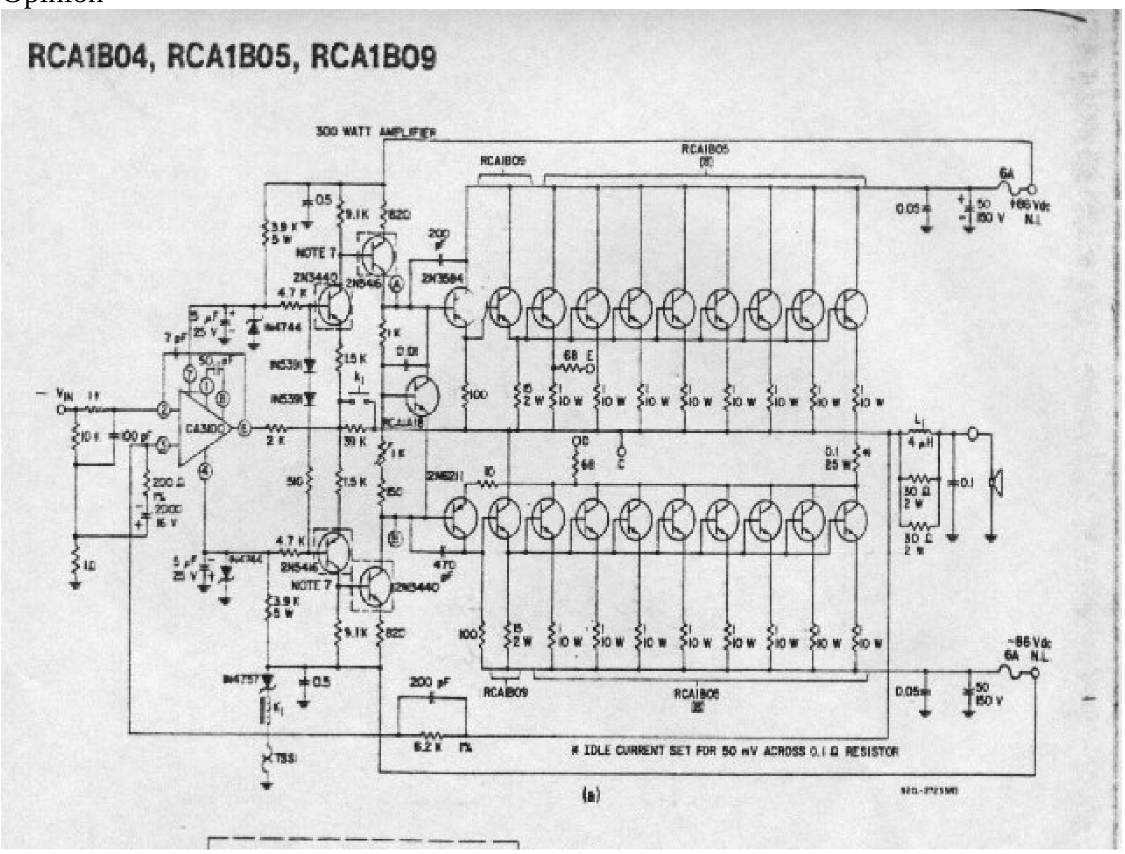
Phase splitter



This type of phase splitter is used in my 40 year old design,Telefunken portable radio with germanium transistors. It still sounds good today, better than most 'modern designs'. Another good trick is if you have a secondary center tap, you can bias both sides of the output stage through it, and not have to add extra resistors.

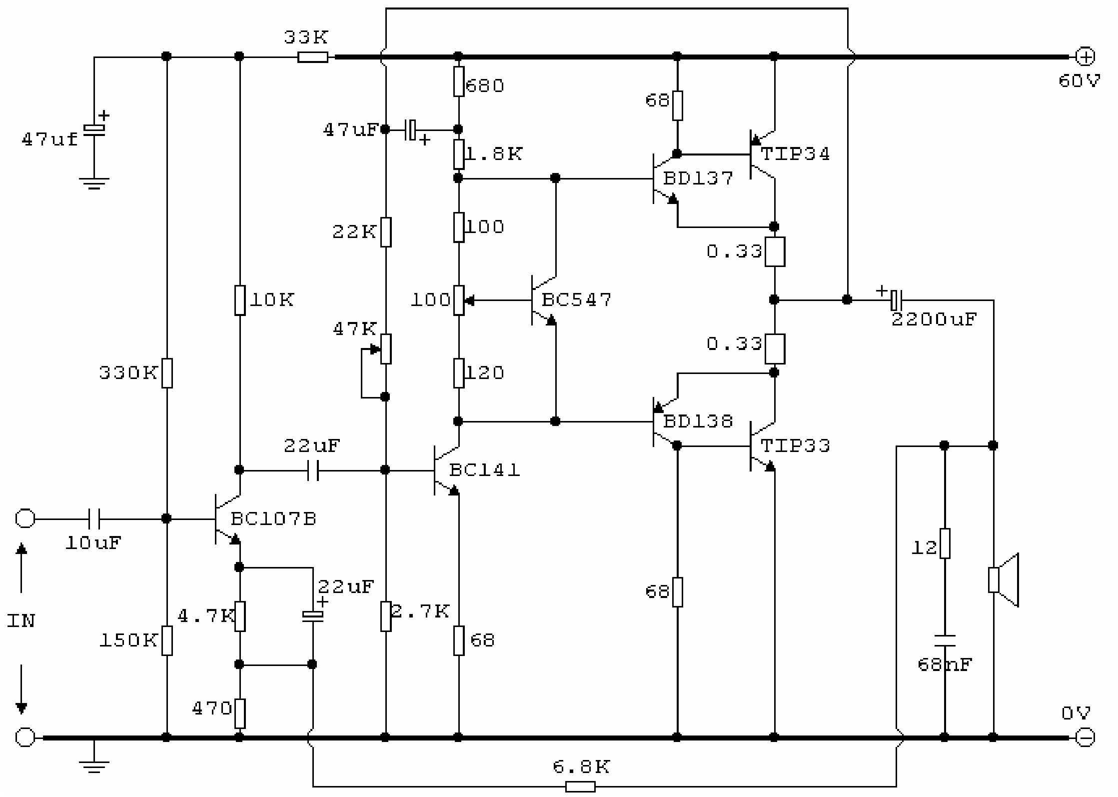
.

Opinion



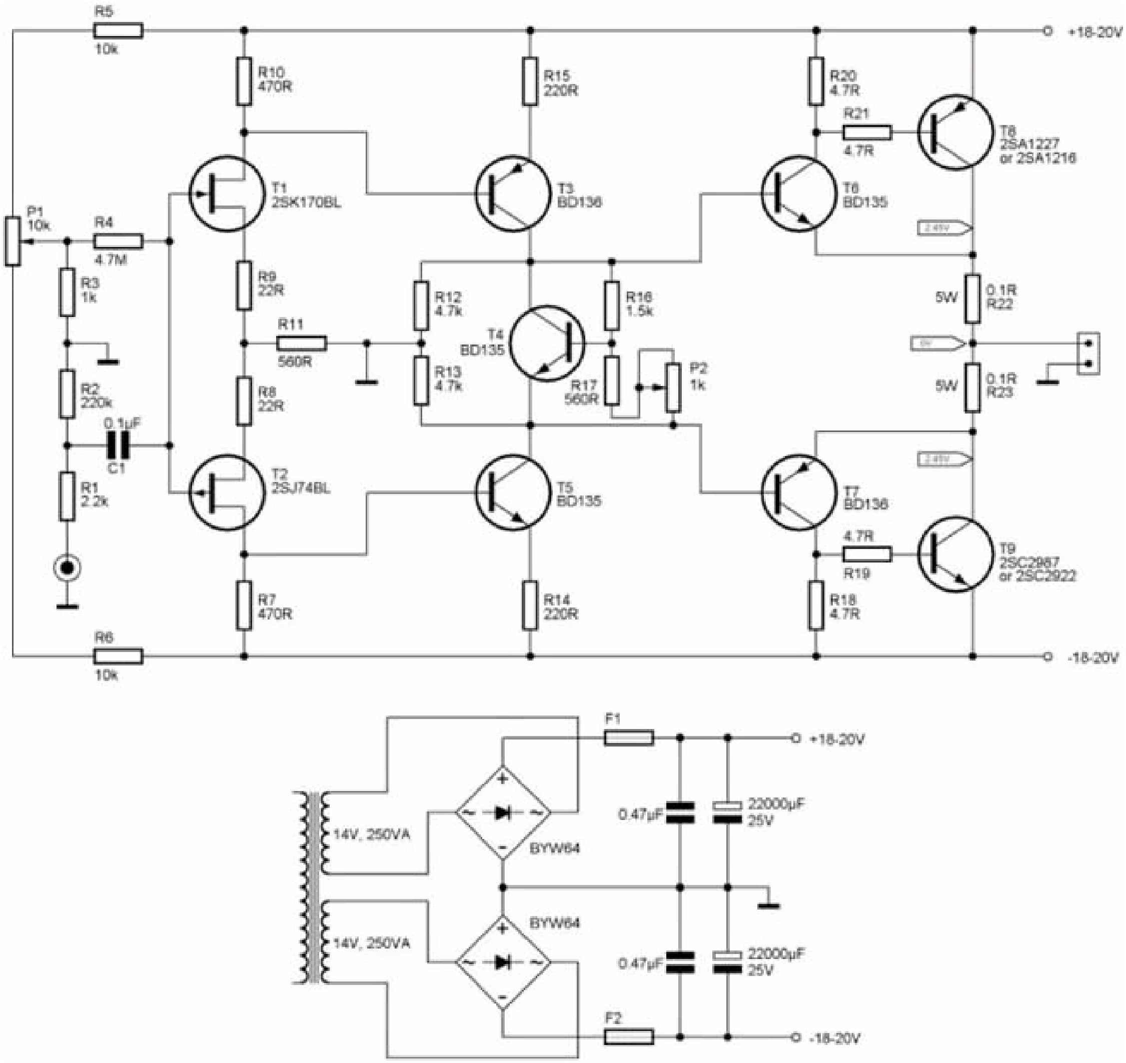
Competent, but very old fashioned design. Kind of like an old Ford motor (pre 1953). You could build it, but why?

Opinion



This design is similar to what we used to design in the middle '60's. This was because pnp devices were much more expensive than caps and npn devices. It is best to direct couple where possible, so using a pnp device for the second stage removes the internal coupling cap, and makes the design better.

Opinion



schematic is not optimum. I have made circuits like this 30 years ago, but they are too hard to bias. The feedback pair will just slow the amp.

The bias control is primitive, and is a step backwards from the one on my schematic. I hope this is enough for contemplation.

Otala amplifier

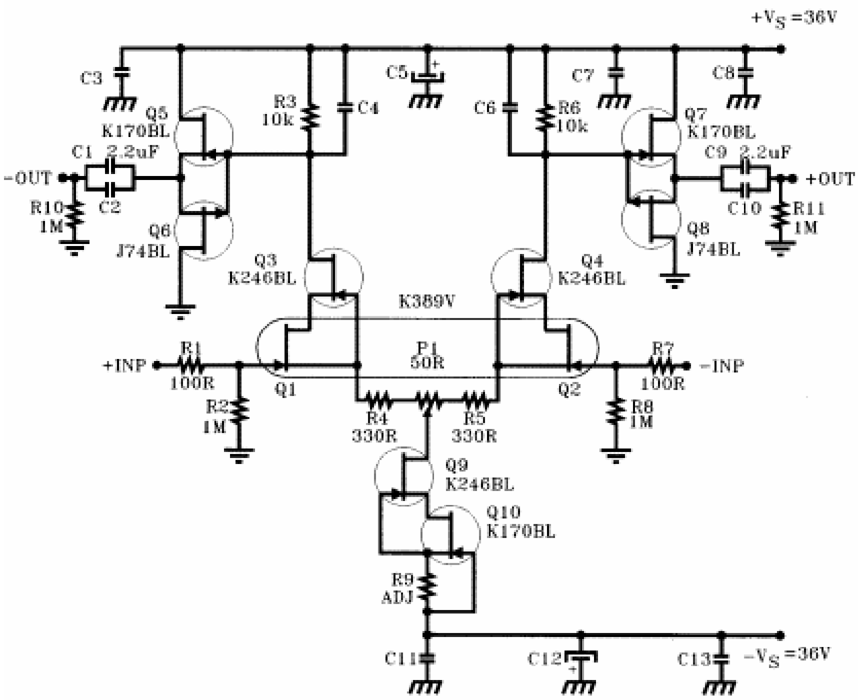
I met Jan Lohstroh at Phillips Research Labs 30 years ago. Apparently both he and Matti Otala developed this amp design, but Matti got virtually all the credit for it.

I bought the very first prototype of this amp made by Per and his friends, after they demo'd it to me in Switzerland in 1975. I kept this amp for 15 years, until it was destroyed in the firestorm in 91. It is an especially good amplifier. A few years ago, I found one locally that cost me $200. I use it in my lab, today, and it still beats my cheaper designs.

How to mod a parasound hca 2200 mk2

First locate any bypass caps that seem 'tacked on' or paralleled with other similar caps. Just remove them, except for one 0.1 uf or so cap in each board location. Any bypasses across power electrolytics should be removed.
Second, make sure that your feedback resistors (47K) are 1/2W Resista or Holco (old), nothing else will do.

Opinion



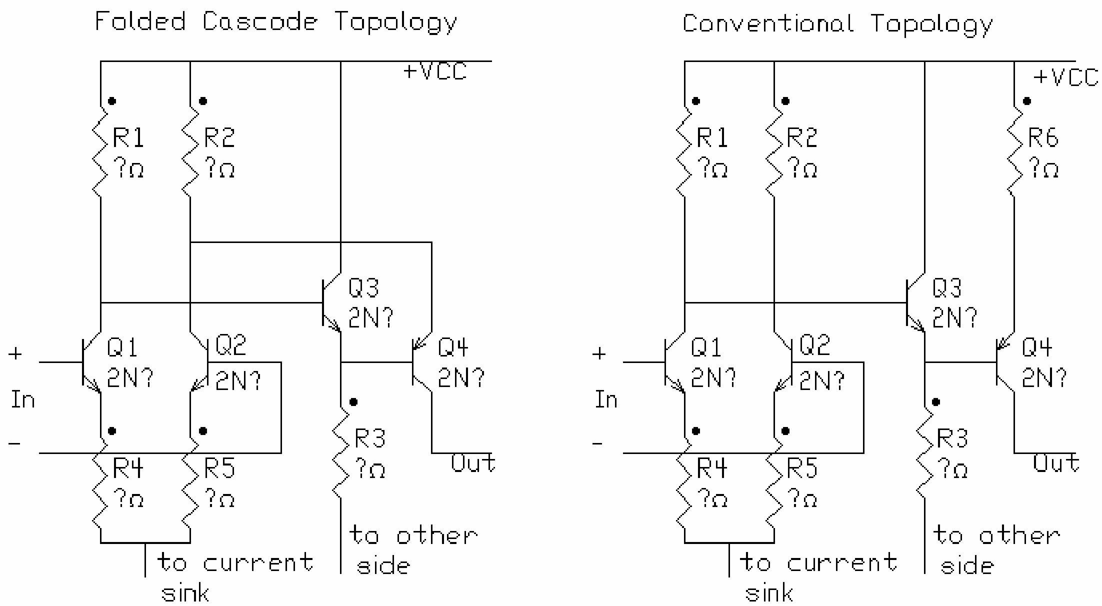
SE BALANCED JFET LINEAMP
EB-2000/402

I built a nearly exact example of this design for the Grateful Dead at Alembic Inc in 1973. It is somewhat dated, but Walt Jung's power supplies should work OK. Be careful about power supply noise, as it can easily get into the design.

replacing some old power smoothing capacitors in a Marantz Model 15

It should not matter that you use a bigger cap. Bigger capacitance is better. I had the mono version of this design in 1969. I went back to designing my own power amps after owning this design. Measured well, however.

Folded cascode (?)



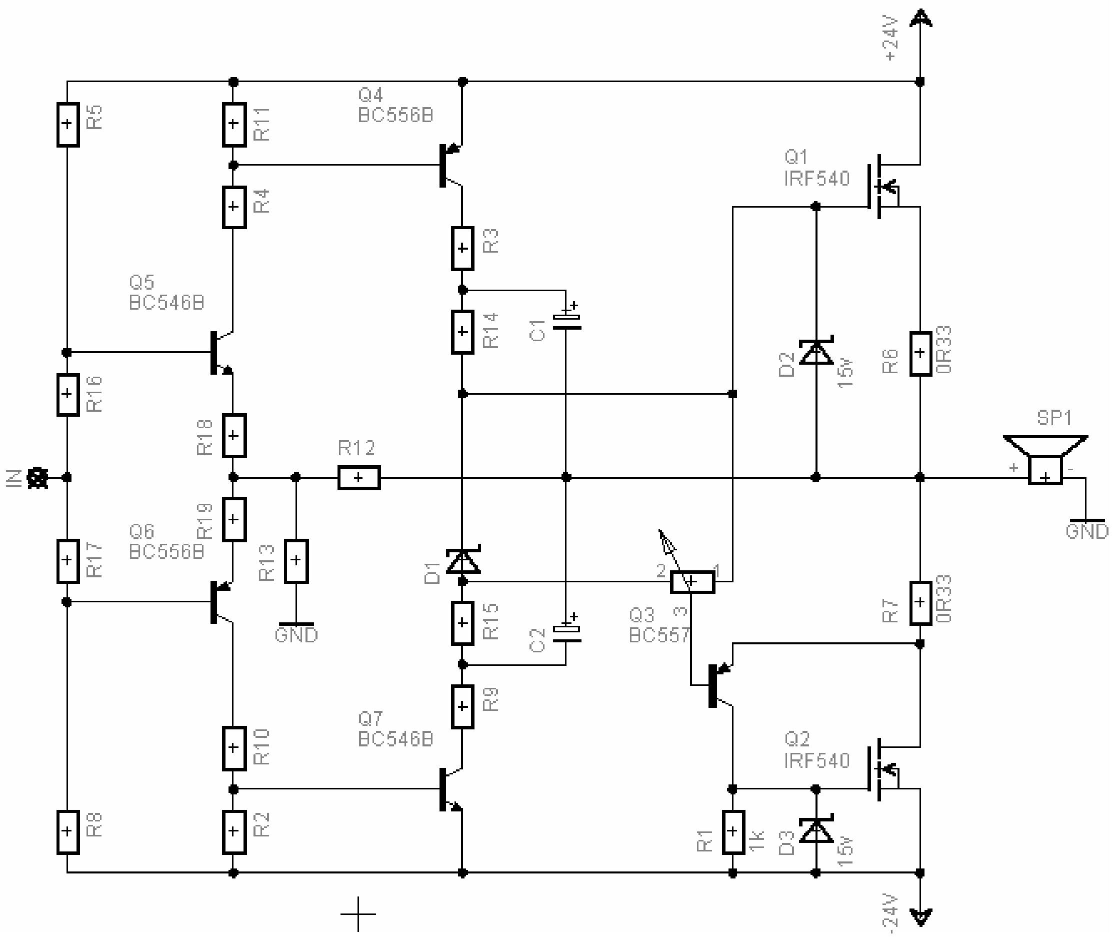
I have used that connection since 1973 or even before. The folded cascode connection is mainly trivial, but it doesn't appear to hurt anything and probably gives a little more gain. The way to test this would be to make a Spice simulation and then add a very large cap from R(1) to ground. Compare with and without the added cap.

Of course, ONLY a folded cascode connection would lower gain, BUT with a normal base connection to the second stage as well, will overwhelm the folded cascode connection with added gain. This is the condition initially stated in the schematic published earlier on this thread. Overreaction like this, with a circuit expert is not recommended in future

tweeter amp

Personally, I would use tubes, but a good class A, fast, low power transistor or fet amp would be next best. You should keep the output power low, in order to protect the tweeter, and optimize the quality at 1W or less.

Opinion



This type of input stage has one potential advantage. It is that the input stage can 'source' more current than just 2I(q), which is the limitation of using a current source for a single or dual differential input pair. This can improve slew rate in many circuits.
The standard transistor input complementary differential input stage was independently developed by Jon Iverson (Electroresearch) and myself (Ampex) in the late 60's, but was

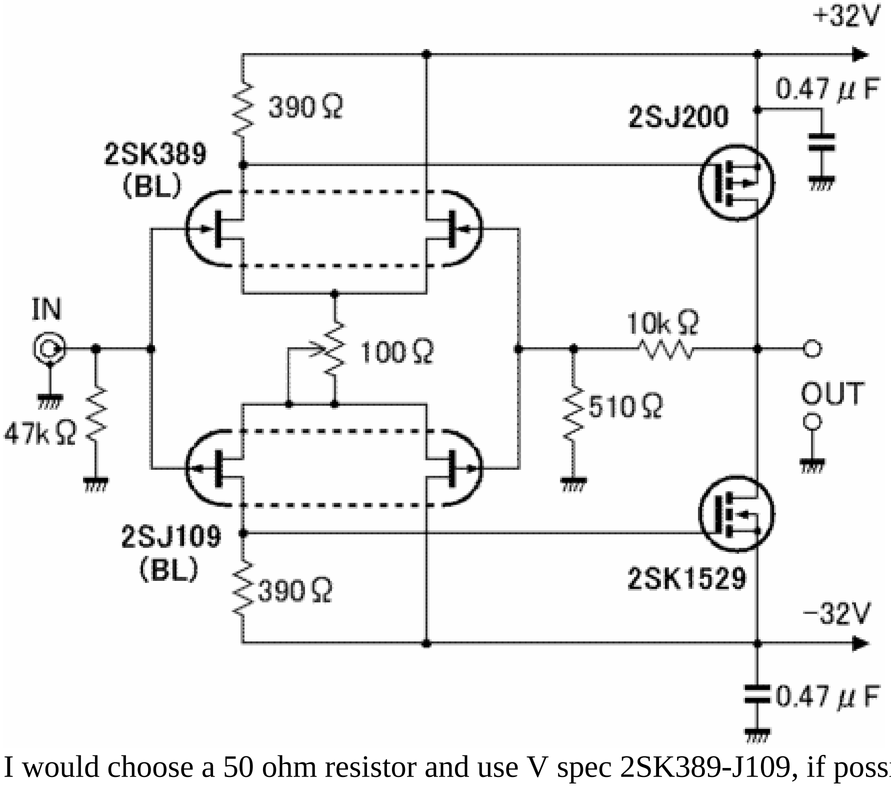
not published. Daniel Meyer (Southwest Technical) was the first to publish it in the early 70's in 'The Audio Amateur'.
The FET input version was developed by me in the early 70's and first published in 'The Audio Amateur' in 1977 as part of the JC-2 schematic.
The 'diamond differential' was independently developed by Sansui in the early '80's and essentially had a single differential input stage. The same circuit can be made with fets in the complementary stage and the diodes eliminated from the circuit.

IF you have to use transistors only, then using 2 current sources to bias the input stage, coupled with a 100 ohm resistor between the emitter pairs would give better performance.

Je-990 op amp

It is a pretty good discrete op amp, but somewhat dated. Fet's on the input 'improve' the initial design.

Opinion



I would choose a 50 ohm resistor and use V spec 2SK389-J109, if possible. This input capacitance of the second stage is pretty large, so higher current drive from the source devices is necessary.

In my opinion, many here are IGNORANT of what is being offered in the audio marketplace as 'improvers', and think it all nonsense, because you have not even attempted to understand what is being sold to the audio public.

Design for living

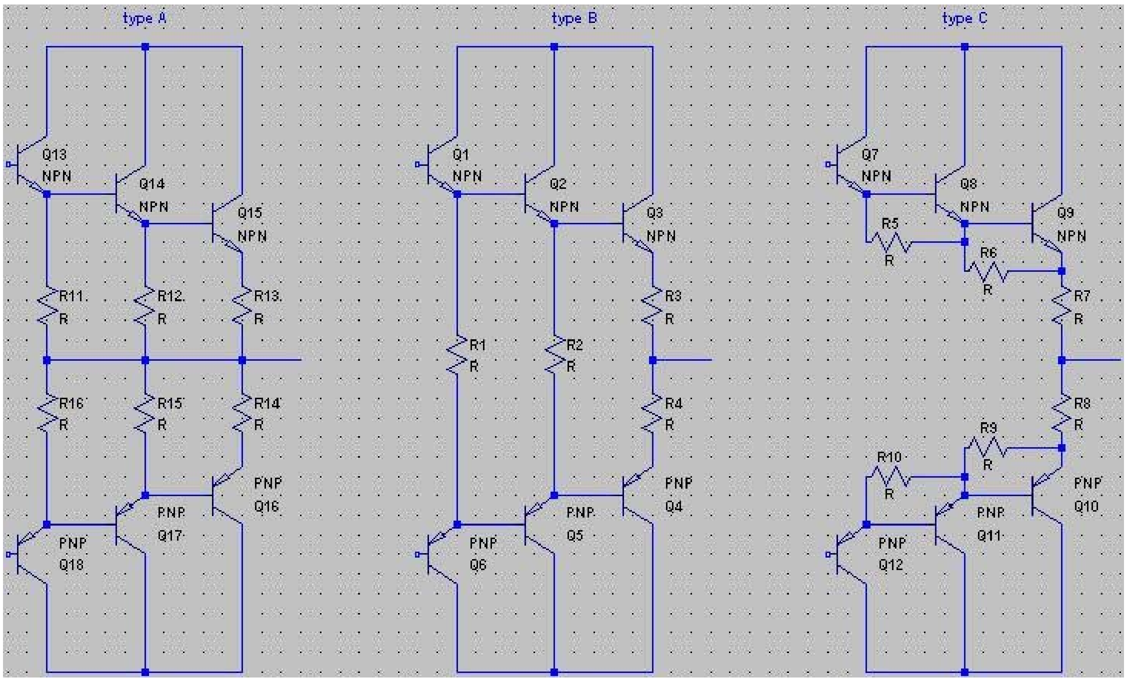
My problem is that I have to make SUCCESSFUL audio products for a living. This means that even if I already know how to make a 'LOW DISTORTION' amp or preamp as well as most people, some of my designs are still considered 'DOGS' to my friends, and they will not use them.

Electrocompaniet amp based on Otala design

It sounds GREAT! I bought my first one, 30 years ago. I have another in my lab sound system.

It beats one of my cheaper designs. This is pretty darn good. Of course, I make more expensive and equally, if not better designs. It HAS a very high open loop bandwidth, and high slew-rate. This is what makes it great, it should beat many commercial amps out there at moderate levels. I cannot compare it to other commercial units, as this is inappropriate.

Opinion



I think that all 3 types are potentially useful. I would have to do an extensive Spice simulation of each, (all else being equal in the semiconductor operating conditions) to see the advantages and disadvantages of each. I have used both A and B with good results, but C looks OK as well, perhaps, maybe even better in some conditions.

FM distortion

Also, let's not try to merge FM distortion with AM distortion. It doesn't prove much, just like equating Doppler distortion and AM distortion in loudspeakers. They are actually different mechanisms, even if they share certain characteristics.
This is phase 2 of Murphy's law. 'It exists, but is not important'

Dynaco 120

The Dyna 120 is a waste of time. It was made too early, by engineers who did not know transistor design very well. Even the Dyna designer who I talked to in 1965 told me to stick to tubes.
Folks, you must understand, marketing pressures like: 'Perfect sound forever', 'No output transformer', etc, force companies to change their designs, sometimes before the new technology has caught up.
I did use a Dyna 120 as a motor drive amp, but that is another story. Sorry, but the Dyna 120 is only good for driving motors, not loudspeakers.

Folks, the Dyna 120 was a DOG! I don't say that often, BUT it came out very early, before complementary symmetry and any circuit sophistication at all. I was told by a Dyna engineer to stick to tubes, at the 1965-66 Hi fi Show in San Francisco, but he may have referred to the PAT4, which was introduced about the same time. I can't be sure after almost 40 years.
At that time I owned two Dyna mk-3 tube power amps and I used a Dyna Pas3 tube preamp, until I replaced it with a Levinson JC-2. I certainly stood behind the Dyna tube stuff, but solid state started off badly, for two essential reasons: First, the transistors were too expensive at the time. Two, the resident engineers did not know how to design with transistors very well yet. Probably one of the first really successful solid state designs in that era was made by Harmon Kardon. Yes folks there really are very poorly perfoming designs sometimes put into the audio marketplace. This has little or nothing to do with taste or preference.

single transistor amplifier

I would try a single N channel VFET, made by Sony, Yamaha, or even a mil supplier, where they are called static induction transistors. This is a natural, high current triode with an 8 ohm loading capability. Unlike a follower, it will have some voltage gain and be somewhat more free from power supply loading. For the record, you will need less inductance as you increase the bias current, so a 1-2 amp idle current would need 1/10 as much inductance as 100ma idle current.

The Vfets that look promising are the 2SK60 and the 2SK70. These will most probably be rare and expensive, but they would be an interesting device for a simple amplifier.
Also, I was confused of the position of the inductor. I have never used an inductor as a load, and of course, it should be very high in inductance if possible. I was thinking about the use of inductors in a choke input power supply.

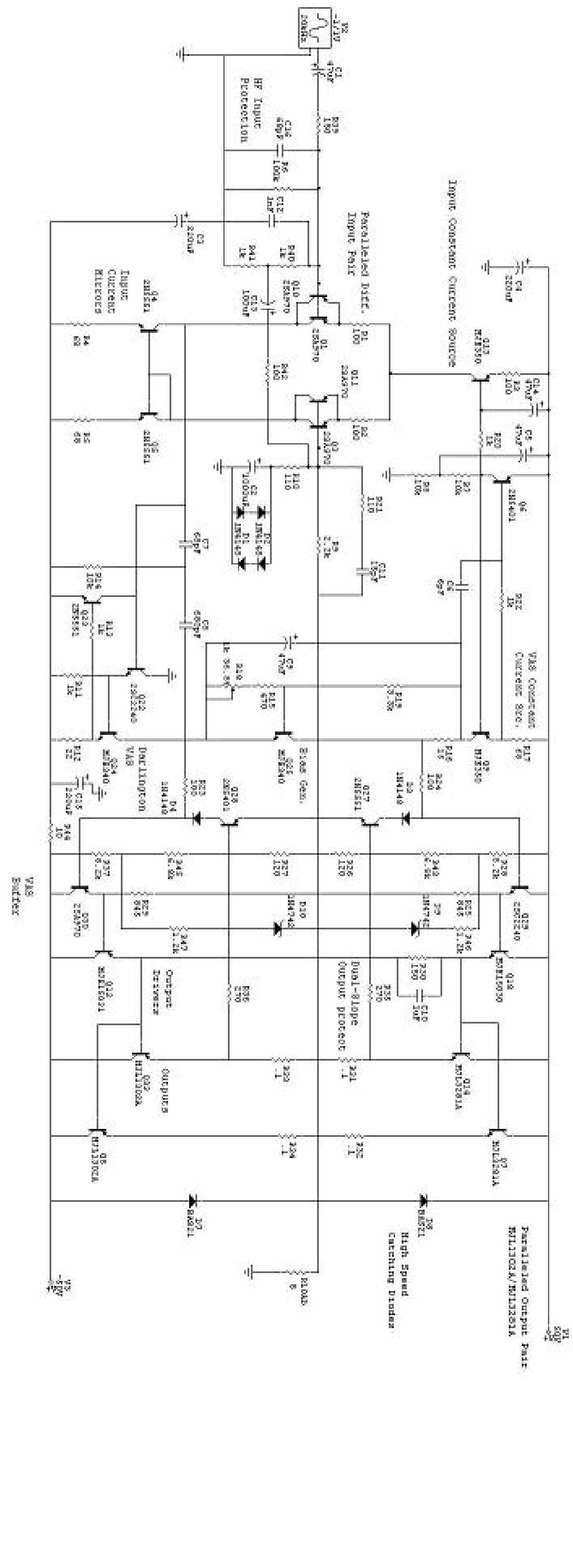
The 2SK60 and 2SK70 are Vfets and are depletion mode devices to the best of my knowledge. I have a few 2SK60's within reach, and I might just put them on a curve tracer to be absolutely sure.

There is no question about it. The 2SK60 is a depletion mode device. I checked it realtime on a TEK 577 curve tracer.

For some reason everyone is confused about these Vfet devices. They are depletion mode, by my definition at least. You have to turn them off to control them. The proof is in the fact that they are modeled with a diode in the gate, just like jfets. Forward bias will reduce the gate impedance to a very low level. I did use devices such as this in the mid 70's when they first came out. They are very difficult to use in conventional circuits, and apparently difficult to produce cost effectively, so the Japanese manufacturers just pulled the plug on them. For a simple, transformerless one device power amp, they would be almost perfect, because they have a very low output impedance without using loop feedback, and the varying output impedance could be balanced against the transconductance to obtain a relatively linear transfer function.

My customers

I would NEVER buy a $15,000 amplifier. I could buy a cheap new car for the same basic price, and have done so in recent years! However, if I were a relatively wealthy American doctor or lawyer, I would invest my extra income, (derived from the pain and suffering of others) into expensive hi fi. I would buy a pair of Wilson WAMM's, the very best digital sources , the best phono turntable and cartridge, and the best analog electronics.
These are MY customers. Not people like me.
Now, what do I get out of it? I get the challenge and joy of making very high quality, cost not important, products. My associates and I get the satisfaction of each and every refinement to the sound quality that we can invoke. I can hear the difference between my latest design and the award winning design that I made 20 years ago, and also the best design that made 30 years ago. I can hear the progress, as I have learned to pay more attention to details, layout, and and actually remove global negative feedback from my preamp circuits.



I see two obvious problems: First the input resistor will compromise the noise so drastically that you don't need to parallel input devices, unless you use lousy ones. Second, the current source load degeneration resistors have to be increased well above 200 ohms, in order to reduce this contribution to the overall input noise. Hope this helps.

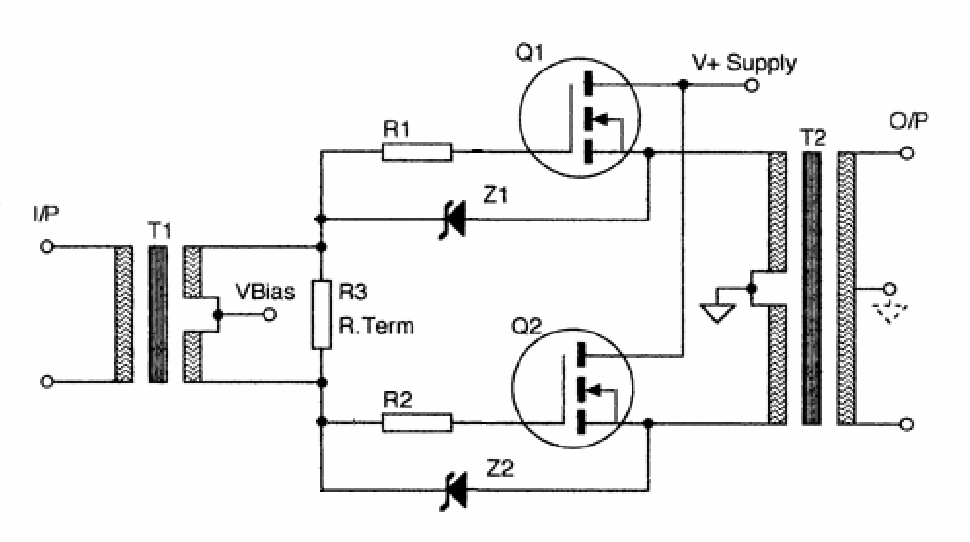
You must use greater than 100 ohms in your current sources in order to reduce the noise gain of the current sources. You will lose 3dB noise from the input stage at least. Is it really worth it? This is grad school level design, but it is correct.

OP-AMP without an output stage

This is normally called a tranconductance amp. We have been using them in audio, at least 32 years, with the JC-2 line amp as an example. They can work very well, so long as the load is not too low in impedance. Be careful with slew-rate. Remember dV/dT= I(out)/C dV/dt is slew rate.

Noise Reduction for analog tape

No matter what you do, extra signal processing will compromise analog recording. Nothing has worked consistently yet.



This design is almost exactly the circuit with germanium transistors that I have in my almost 40 year old Telefunken Bajazzo portable radio. Just used it today. Sounds great! PS Dick Sequerra (hi end guru) was a consultant on that project. This portable radio runs rings around ANY portable component that I have ever heard, including Class D portable circuits. I have owned 5 or more! Bought my first in Germany, 1965. Second in Berlin 1967, etc Found this one in a pawn shop in San Francisco, CA USA in 1972. Gave it to friend, and he gave it back to me when it developed problems. Fixed it, and I won't give it back. I misread the circuit, BUT I do have both the driver and output transformer, AND same type transistors on the output. For some reason, it sounds pretty darn good. Maybe lack of too much negative feedback.

this design would not be optimum for RF, BUT I once built a similar RF amp with the features you want. Please consult an Amateur Radio Handbook for instructions on how to make a good RF amp. It is completly different from audio, and even my fastest designs would make lousy RF amps, even though I could probably run AM radio through them easily enough, with a little adjustment.
You can still use dual transformers, BUT the advantage of high input impedance is lost above a few MHz, because of the high input capacitance of the fets. Common source pp design will give you more POWER gain, which is what you need.

5. Parts

Output devices

An objective appraisal of different components will show different measurements; some are better, some are worse. This is true for both caps and transistors, as well as fets.
For the record, Sanken power transistors are fast, linear and have a good safe area. In reality, and sometimes on cheaper products, we often use Toshiba transistors. Both companies easily outperform similar American products. I wish this were not the case, but I am not fool enough to deny it. With caps, there are differences in construction that make a difference with top performing brands. Of course, your standard Sony, etc will not get as much improvement, but state-of-the-art preamps and power amps need good caps as much as they need good design. I know, because I design at all levels of price, and the cheaper stuff has the same design excellence, yet does not sound as good as the more expensive stuff. If it did, I would be very happy, but audio reviewers and audiophiles give me enough feedback to know that this is an accurate assessment of the situation.

Those of you who care, should get AN1308 from On Semi (MOTOROLA) and see what output transistors they recommend. How about: 2SA1302 and 2SC3281 from TOSHIBA. The MJL devices are apparently copies of the TOSHIBA devices. Perhaps built under license. Why don't you compare the specs between the two brands? Go ahead, test next week.

Sometimes we even use Motorola, but again, a different part pair. We use Sanken for our best equipment, because they are bigger and better.

In the old days, 35 years ago or so, we used devices with something like 4MHz F(t). Sometimes, people used even slower devices like the 2N3055, or the 2N3773, or even the MJ15003, that had an even lower F(t). With a 4MHz F(t) and a first class driver stage that looks even more up-to-date than the schematic in the Motorola app note mentioned previously, I could get 100V/us with Motorola 4MHz devices. With 2MHz Motorola devices, I could get only 50V/us and twice the distortion at all levels.

In 1980, the Japanese released new devices that had over 20MHz F(t). Then I could achieve about 500V/us with the same circuits. Now I am not claiming that 500V/us is absolutely necessary, but we instead were able to remove the 2UH output coil and do other things that the lower F(t) devices did not allow us. To alternately use these very different output devices in the same exact design, is surprising to say the least.

Parts

Still, just because a certain brand of resistor works in one location, does not mean that it works everywhere, sonically. I don't know why, but this understanding works for me.

Solder quality

Get some good SN62 or SN63 solder. Get a NAME BRAND, not Radio Shack. One lb. will last you forever, and you will always get good solder joints. Don't cheap out here, as you will just make your life (in soldering) that much more difficult if you do.

IF you are new to soldering, just get SN62 or SN63 from a quality manufacturer. Don't waste your time, at this point, with anything else. Trust me, I built my first Heathkit in 1959. Some kits that I built, even then, still work, but they look pretty bad, inside. SN62-63 will keep you from cold joints and both are low temperature solders. Flux quality(inside the solder) is important! Don't cheap out with RS. Get a name brand.

Multicore is good, RS is bad.

Components

For example, I don't know WHY pointed feet should sound better than rubber feet, but many of us find it to be true. Also, many of us find that we often 'agree' on the sound quality of a certain component, even if we have never met each other.
For the record, I have thousands of mil spec components that sound LOUSY, in audio products. Want to buy some? I can give you a good deal! This is your chance to prove me wrong.

Silver wire

Just to offer a little personal experience to this: Silver works great, IF it is pure and is broken in before installation. IF NOT, it may not be worth your while.

5.1 Transistors

MAT02 and LM394
he silly part of all this, is that you folks CAN'T read a data sheet. There are differences between the MAT02 and the LM394, BUT it is not noise. Also, if any of you actually bother to read these data sheets, please NOTE the very high capacitance shown in the

Ccb and Cbe graphs. This is as bad or worse than that of an equivalent noise FET, like the 2SK389.

Both data sheets were listed on this thread. Both data sheets are complete, as far as they go. Neither device is 'better' than the other, for audio, yet a 2SK389 will beat both for low noise over a range of inputs.

You might actually find more 3rd harmonic generated with the fet input, BUT it is a CLEAN 3rd, just like analog magnetic tape. The advantage of jfets is that they STAY QUIET over a range of source impedances, including both resistive and inductive. They don't require much input bias current either, SO you can leave out the input coupling cap. This is important! You can also use, with ease, MUCH more idle current on your input stage. This makes driving a comp cap easy, and potentially increases your slew rate. There is much more, but enough for now.

I meant FET's used in a diff pair input. The 2SK389 is a dual fet pair, and is mostly used for differential inputs. Differential input, by definition has NO second harmonic, if everything is matched.

FETs

Fets are now, and have been for the last 25 years, as quiet as bipolar devices in almost every application, if you use Toshiba input fets. Bipolar transistors, because they have somewhat higher Gm, can have slightly lower short circuit noise, but at realistic impedance levels and currents, they are usually just as noisy or noisier than discrete fet input stages.

Matching FETs

BUT it is fairly easy with just a multimeter. You just short the gate to the source and add a 9V battery from the drain to the source. This gives you Idss. But how to measure? Put a 10 ohm resistor between the drain and the battery and measure across it. 0.1V=10ma and so forth. Matching Vbe is much more difficult.

monolithic JFET pair with the DC precision

That is easy, a 2SK146 will do the trick. It will also keep low noise at all reasonable impedances, up to 1 meg ohm or so. Of course, these devices are not easily available, so you could parallel a pair of 2SK389's and get the same performance. I work at 0.4nV/rt hz with complementary paralleled fets in my Vendetta Research design. I have achieved a 10 ohm overall equivalent noise level with this circuit for the last 20 years. Before that, I used bipolar transistors to achieve the same result, approximately 30 years ago, with the introduction of the Mark Levinson JC-1 pre-preamp.
The problem with transistors is their NOISE CURRENT which is very high when the NOISE VOLTAGE is low. It is virtually impossible to get both very low noise voltage

and noise current with a bipolar device with an inductive source, such as a tape head or a moving magnet phono cartridge, and they can be very noisy with source impedances above 1K ohm or so. Fets can be operated at higher currents for low voltage noise, and still have almost unmeasurable current noise.

As far as noise is concerned, in this case, fets can easily keep up with bipolars.

2sk170BL or 2sk369 for MC stage

It's been pretty well discussed, but the 369 is the 'replacement' for the 2SK147. This part should be 3 dB quieter than a 170, but two 2SK170's in parallel essentially equal a 147, in both noise and input capacitance. Please, let's not quibble over small differences. The biggest concern should be if the devices have very low 1/f noise, as some Toshiba processes can be more noisy than others. This will be very important in a MC phono stage.

J112

J112's (American type) are analog switches that can be used for current sources, switches or cascodes. Their Idss can be just about anything above 5ma.

using the 2N3819 JFET with a 40v supply

You are exceeding the best working point of the 2N3819. RF fets, like the 3819, will leak excessively well below the max voltage point. This can cause noise, and make biasing difficult. A self cascode with the cascode device's gate connected to the source of the input fet, and the cascode source connected to the input fet's drain will work OK, but you should use a more modern fet, that one is almost 40 years old! I recommend a 2SK170 or a J113 (american type) fet for this voltage range.

High voltage (>60V) P-channel JFET

I doubt if there is anything available. Usually, P channel is lower voltage than N channel, all else being equal. Cascode with 1/2 A Hitachi mos fet like the 2SJ79 gets you up to 200V. Works great, been using it for more than a decade. Other, low current, mosfets will also work.

Old JFET in mic goes noisy

Clean with pure alcohol, pressure air dry, before making changes. 2n3819 fets are easy to get and usually work well. Important resistors are VERY HIGH in value, not normally available.

Stick with the 2N3819. I know from experience. Yes, there are 'better' fets, but I doubt that they will be functionally quieter than a selected 3819. Watch out for input capacitance. Low noise fets have more.
Ultimately, the VALUE of the input resistor(s) will set the noise of the mike. Higher values like 1 gig ohm are best. 1*10^9 ohms or higher are difficult to find.

Borbely parts

Please don't waste your time trying to change the parts. They are not easily second sourced. Buy them from Borbely. They will be relatively expensive, but you are in no position to find them much cheaper. In fact, just get a 'kit' from Borbely, and don't waste your time trying to make it on the cheap. You will pay more, in the long run.

I understand now that the cost of these components is very high, especially from Erno, with the rate-of-exchange. I was trying to save others, who might live in Germany or many other places in Western Europe, to not waste their time trying to save a few Euros, but I will tell you the whole situation, in order to help you further.
We have used these devices, that Erno Borbely designs with, and sells, for many years. Once, I had 3000 2SJ79's in my possession along with the complement, the 2SK 216 (I think) . They were burned up in a firestorm, but if I had them today, I could sell them and buy a Porsche with the money. What used to cost 1 euro each, 20 years ago, now may now sell for many, many times more. When you are trying to buy just one to 5 parts, the cost of your purchase is always high, because of the handling cost for the distributor. After all, 1 piece, or 100, or 1000 pieces takes the same processing time. If you can find a cheap distributor, then you will have had better luck than I have, over the years. Give it a try, if you want.
Second, these devices are SOTA, which means the best available, like Porsche or Mercedes. These parts will always cost more than a near equivalent part that is not quite so unique. I cannot personally recommend other parts, because I do not use them, but there are somewhat similar parts available at a reasonable price.

5.2 ICs

Op-amps

First, OP AMPS had been around for almost 10 years. In 1966, I worked with the UA702 and UA709 op amps, and the UA741 op amp in 1969. These parts were a god-send for minaturization and for servo control, but fairly lousy audio devices. In 1970, Harris Semi (then Radiation Inc) came out with a dielectrically isolated op amp with low noise (9nv/rtHz) 50 ma peak current, +/- 24V/us operation, and a slew-rate of +5/-2.5 V/us slew rate. Selected units could measure fairly low distortion as well. This seemed to be the answer to an audio designers needs, BUT once we used them, we found them not to be sonically as good as tubes.
What to do? Well I decided to build a discrete circuit with a fet input that had a minimum circuit thru-path, high open loop bandwidth, and as linear as possible operation for each device.

For line amp operation, the circuit that we have previously discussed worked for me. Both Mark Levinson and the Grateful Dead used this circuit as line drivers for several years.
For higher closed loop gain needs, another circuit design was necessary. Then, the op amp configuration works better.

Let's go on, if we can, as to WHY we would want to build simple circuits, rather than use complicated thru-paths? Now, when I mean 'simple' I don't mean crude, or elementary. Push pull is OK, so is 4 quadrant operation so that you can have both balanced inputs and balanced outputs. This can be useful, even when using only one output, as a phase inverter for generating absolute polarity with different software.
Also, think about distortion and what the harmonics look like.
And finally, think about high open loop bandwidth, which is difficult, but not impossible with OP Amps. Why would we want high open loop bandwidth?

It is interesting that I made a modified version of this design to make the Grateful Dead Line driver, that has to send the stereo signal from the mix board in the audience to the stage, 100 ft or more. This is probably the nastiest load that anyone here would ever encounter.

Transconductance opamps

Actually transconductance amps are MORE STABLE than op amps. This is because, instead of ringing, they become more compensated by the load capacitance.
An op amp would have a follower of some kind. In this case, the cap load is buffered from the compensation, and this creates a separate second rolloff of the high frequencies. This is what causes ringing.
The synthetic inductance is another issue, and an interesting one. I suspect that you could build a 'pathological' transconductance amp that had low gain-bandwidth and low open loop bandwidth that could be problematic. This particular design is high gain-bandwidth and high open loop bandwidth, so it may not have as much problem with synthetic inductance. However, it is an interesting question.

Ron Wickersham, Bear and I had the responsibility for the components of the system. There were others as well. Jon Meyer was a personal friend to me, but he did not work with this system. Later, Jon Meyer and I went to Switzerland to work on another project, and this system was completed by Ron Wickersham.

Another small fact. The circuit that we were originally talking about on this thread that came from the JC-2, was originally designed for the big 'Wall of Sound' system used by the Grateful Dead. Mark adopted it for the JC-2 because of its success with the Grateful Dead in sounding better than selected existing op-amps.

317, 337 Regulators

However, this is my take on 317,337 IC regulators: BE CAREFUL! Don't put a really good cap directly at the output of one. It will cause noise ringing. There was a good article on this in 'EDN' or 'ED' some years ago. What happens is this: The regulator starts increasing its output impedance, because it has a finite bandwidth and must roll off its gain early. This RISE in output impedance looks just like a synthetic inductor at the output. Put a really good cap directly at the output, and it RINGS! This is because there is no series damping. Add a 2 ohm resistor in series with the really good cap, and you are probably OK. However, why then would we put a really good cap at the output, if we have to add a 2 ohm resistor to spoil the Q? This can make things really interesting, and confusing, if you don't have any idea of what is going on.

5532

we have had these series IC tests for decades. As I remember a designer made one about 30 years ago to convince Mark Levinson that IC op amps were not audible. Oh well!
It seems, as we add more IC amps in series, the damage has been already been done with the first IC, and the rest don't count for much.
I have personally never heard an op amp IC beat an open loop discrete design, but I am told that the AD797 is darn good! And in many cases, superior to many discrete designs.

5532 opinion

However, there are other, better op amps that can be substituted. Some have a reasonably high open loop bandwidth, others have, at least, a more linear input stage than is possible with the 5532.

5534 op amp

Anyone who thinks that a 5534 IC is 'good enough' for hi end audio products, makes me laugh! I seriously tested it over 25 years ago, and found it lacking.
The AD797 is a much better op amp, but not perfect for all audio applications. However, I compete with designs using the AD797, and I think it is one of the best IC op amps available. Still, I will stick to discrete designs for my best efforts. Why? Because I can use class A EVERYWHERE in preamp designs, and complementary jfets anywhere I want. I like jfets, they are VERY quiet, have a high impedance input, and have a more linear transfer function. IC's don't have complementary jfets, it isn't practical at this time to put them in. Therefore, Nelson, Charles, and I can do interesting and sophisticated circuits that are not possible with IC's.
Then, there is the problem with thermal feedback. ALL IC'S have thermal feedback, some more than others. This might be MORE important than open loop bandwidth, who knows.
I have the Barrie Gilbert article around here somewhere, but I can't find it just now. I will re-read the section with PIM being discussed, when I find it. Let's find out what the article really said.

JFET vs bipolar op amp

In principle, the AD797 should run rings around the 5534. Please understand, I have used the 5534, since 1977, when I was first sampled by Signetics. It was a serious improvement in audio IC op amp design and performance. However, we found that IF you bypassed the input stage and substituted a fet input stage, we could make a better sounding device. We (my tech actually) made them for Dave Wilson for his $100,000 speaker system's active equilizer. Dave paid $80 each, 15 years ago, I believe.
The AD797 is similar to the 5534, but significantly improved, at least in principle. I'm sure that the designer of the AD797, Scott Wurcer, knows the ins-and-outs of the 5534. He would not have made an inferior device on purpose, at least.

Opamps

It just so happens that the UA709, 5534, AD797, and the AD829 for example, have the same, UNDEGENERATED, dual differential transistor input stage. Why? Because resistors add NOISE to the input.
To me, it is like this: WHEN you have significant open loop distortion generated by the input stage, even if everything else is perfect and does not contribute, (which is impossible) then the open loop bandwidth will be modulated with signal level and frequency. This will cause a dynamic phase shift of the audio output which is outside the direct control of global negative feedback. A dynamic phase shift would not be easily seen with a simple harmonic distortion measurement, at least in my estimation. Therefore, in principle, you could have a low distortion design (AM distortion that is) that actually has a significant amount of dynamic phase shift, when real audio signals are put through it. This MAY be why IC op amps (with a few exceptions) don't sound as good as most discrete designs.
But what the heck, why not use the cheapest op amp that makes you believe that you have achieved audio perfection?

I got my copy of Motchenbacher and Fitchens' 'Low-Noise Electronic Design' when it first came out about 30 years ago. It was the first really good book on low noise design, and its only limitation today is that there are much better parts that were developed in Japan after the book was published.
The technique used by the late Deane Jensen is more easily done by using a low noise jfet pair in the input stage. It gives the same short circuit noise, but reduces input bias current and sensitivity to input loading.

>what do you think the OPEN LOOP BANDWIDTH is for the AD844? What about the AD797?

Maybe, just maybe, there is a clue here for best audio quality.
Yes, I meant the AD797. Almost 40 years ago, I worked extensively with the 2n697 and it gets stuck in my memory.

Page 7 of my data sheet for the AD844 says:"The open loop pole is formed by Rt (2.5Mohm) in parallel with Ct. Since Ct is typically 3pf, the open loop corner frequency occurs at about 18KHz."
For reference the open loop bandwidth of the AD797 is about 100 Hz (or less).

I accidently found my data sheet that was originally sent to me 15 years ago or so. Remembering that YOU liked the AD844, I thought to look more carefully at this data sheet.
To my happy surprise, I found the 18KHz number. From the same file, I also had the AD797 data sheet, so I compared. These are two excellent examples of IC amplifier design. Is there any significant sonic difference between them?

Op amps

I have worked with op amps since they were first designed, 40 years ago. I have ua709's that are 38 years old, in my lab stock. I have used many hundreds of IC op amps in designs, and I test almost every new IC op amp design as they become available. I can still build discrete circuits that are essentially op amp based, that work and sound better than any op amp that I have tested. Why? The only factor that is obvious is the relatively low open loop bandwidth of most of the IC op amps. This is why we are addressing this issue.
If manufacturers can offer a truly superior IC op amp that is not compromised compared to discrete designs, we will use them, almost exclusively, and only use discrete designs for special applications. We want this, as it would save money, time, and space for us, and cost reduction for our customers. So far, even the OPA134 and AD797 are just OK, not quite what we can make ourselves in specialized situations such as buffers and line stages.

IC buffer in headphone amp

If you really want quality, just make a discrete push-pull class A buffer from Fets or bipolar devices. The only other alternative is tubes. Trust me.

Subjective evaluation and op-amps

Successful designers actually LISTEN to their creations and those of others. This provides them with the 'feedback' necessary to see if they are on the right track. Many of us have tried the 'high feedback', multiloop, and other approaches, with some disappointment. That is why we tend toward lower feedback and higher open loop bandwidth. Is it because of PIM? We don't know yet, and neither does anyone else, BUT we know what sounds good, and you will find examples of our designs in Class A listening ranking. We don't get that designation by NOT listening to our design efforts and modifying them accordingly.

I would like to point out that most of you have no idea how the ear hears phase, and when humans are sensitive to it. If an op amp generated a changing phase with every different frequency and amplitude, even if it is small in amount, I suspect that it could be detected by the human ear.

The bottom line is that high end audio has minimized the use of op amps, because we can hear what they do to the music. To do this, we have live music, or at least class A discrete designs, either tube or solid state, of the highest quality to compare to. So far, almost all op amps have come out second best.

Folks I just ran a test measuring harmonic distortion in a sine wave generated by a function generator. I also was able to take a simultaneous fft measurement as well as a thd measurement. Then I added FM modulation to the sine wave. It could be easily seen on the fft, and the oscilloscope tracing instability, BUT it took a lot of FM to change the thd measurement. This means that the thd test is relatively insensitive to FM modulation. So there!

as I have never studied FM modulation formally. However, what I did find was that at a large range of modulation frequencies, I could get results on the FFT, but not with THD. This makes sense, since the THD notch is what can't follow the change in frequency.

have worked with op amps since they were first designed, 40 years ago. I have ua709's that are 38 years old, in my lab stock. I have used many hundreds of IC op amps in designs, and I test almost every new IC op amp design as they become available. I can still build discrete circuits that are essentially op amp based, that work and sound better than any op amp that I have tested. Why? The only factor that is obvious is the relatively low open loop bandwidth of most of the IC op amps. This is why we are addressing this issue.
If manufacturers can offer a truly superior IC op amp that is not compromised compared to discrete designs, we will use them, almost exclusively, and only use discrete designs for special applications. We want this, as it would save money, time, and space for us, and cost reduction for our customers. So far, even the OPA134 and AD797 are just OK, not quite what we can make ourselves in specialized situations such as buffers and line stages.

Thermal feedback in opamps

I agree with Walt Jung that 'thermal feedback' can be as important as open loop bandwidth.

???op amps and open-loop bandwidth

null' testing is interesting and informative, but the residual, whatever it is, must be evaluated with a spectrum analysis, if you really want it to be meaningful. You can try it, of course, but usually LINEAR DISTORTION, do to phase shift, rather than nonlinear distortion, will dominate and limit the test.

I really wish that it did work well, BUT we (Walt Jung, Scott Wurcer, and I) found that even a single capacitor can have a measurable residual when directly compared to another capacitor of similar value, but slightly different material composition. How about an entire amp?
It is true that the apparent audible null is convincing, but why is it that the Hafler 280 is not the truly representative amplifier for the audio industry?
Actually the 280 is very similar in concept and execution to the first Levinson power amp, many of the Krell amps, and the Parasound amps. Yet, each and every amp sounds somewhat different. Why?

I might suggest that it is the higher order odd harmonics that are of most concern. You should match your devices, just because we normally would use matched FET's or bipolars in quality designs, and the resultant 2'nd harmonic from any mismatch potentially obscures the resolution of higher order harmonics. For the record, we can happily listen to a great deal of added third harmonic distortion as well as 2'nd. This has been shown by listening to decades of analog tape recording that have typically 1-10% third harmonic at 0 Vu and higher levels. 0.1% harmonic distortion is virtually un-noticable, but don't think that almost any amount of 7th harmonic is! Also, you might consider noise, when you add resistive degeneration to bipolar transistors. Fets will be lower noise, and have almost no bias current to deal with as well!

IF you degenerate any device enough, it will not behave as it would without degeneration. It is true that a really starved FET will have the same distortion profile as a bipolar transistor, BUT this is not a good place to operate the device for analog audio.

I have to agree that the fundamental equations used do describe the behavior of tubes, bipolars, and probably fets were developed from statistical analysis. Engineers are usually just given these equations (like Child's law) or the derivation of Gm in bipolars. Fets tend to be a degenerate form of bipolars, unless operated at unrealistically low operating currents when they merge into bipolar Gm behavior with current. In fact, most physical processes seem to behave in a similar way. Too bad that I hate statistics.

It has been my experience that negative feedback is something that I prefer to minimize and eliminate, if possible. There is another type of distortion that might be important, that has not been discussed. This distortion was found by Dr. Hirata and uses a special test box to evaluate. Perhaps we should analyze this distortion mechanism as well. Hirata's papers can be found in the 'Journal of the AES' around 1980. Perhaps someone can find or provide a link.

Actually there is no clear answer. For example, Halcro makes an amazing amp that measures VERY LOW in distortion, DUE to lots and lots of negative feedback. I make an amp that measures pretty well, BUT not as well as the Halcro in harmonic distortion at just about any level. I use a moderate amount of negative feedback. We both have a Class A rating in 'Stereophile'.

What is the advantage of high negative feedback?

My first experience with open loop bandwidth involved a comparison of my own Comp Diff Class AB1 design (.5A bias) compared to a vacuum tube triode amp in 1969. Both amps were rated about 10W and drove a Klipshorn with 104dB/W sensitivity.
Both amps measured essentially the same closed loop bandwidth, IM distortion at various levels, ie less than .005% at 1W, slowly rising until clipping. Both had 2'nd and 3'rd order distortion, almost exclusively, both were stable with any capacitive load, and both had a damping factor of about 30.
We still heard a difference. Why? How?
Well the only major difference between the two amps, as I saw it then, was that the triode used 20dB of negative feedback and my amp used 40dB to get the same results. I can presume from this that the open loop bandwidth of my solid state design was about 10 times less, since the closed loop bandwith was the same in both amps of about 100KHz.
Otala's paper, gave me a better understanding of why the amount of negative feedback mattered, more than 1/3 of a century ago.
I don't always agree with Dr. Otala, but his input has been more useful to me, over the decades, along with Dr. Hawksford, than almost any other designer.

LT1166

I tried to introduce the LT1166 into our multichannel amps. We had BIG problems with high frequency distortion and we could not meet THX specs. We had to drop it. Still, in principle, it looks pretty good.

5.3 Resistors

Resistors

I don't know how to prove this to you, BUT I have hundreds, if not thousands, of resistors that are mil spec, beautiful, and with gold plated leads. Many have a clear 'glass' case so that you can even see the resistance element. I made a preamp with these resistors about 25 years ago. It was a disappointment. Later, I made similar preamps with 10 cent, 1% Resista metal film resistors, and the results were spectacular! I won several awards. Want to buy some of my gold plated mil spec resistors? I will sell them cheap!

Resistors

My favorate resistor costs 10 cents American, can't you afford that much? The important thing about silver wire is that it sounds different than copper wire. It just does. Tough nuts!

We have found that Resista resistors work very well for even the highest level audio electronics.

5.4 Capacitors

Dielectric absorbtion

DA might have a non-linear component. This was measured by Cyril Bateman in his cap distortion articles in 'Electronics World'. Hysteresis itself, I think is a separately measurable nonlineartity, at least this is obvious with cheap ceraminc caps.

Capacitors

If you want the 'best' then only teflon will do, next is polystyrene, then polypropylene, then polycarbonate, then down the road, is mylar. Significant DA is the most important difference.

Feedback capacitor

We usually use a polarized electrolytic cap as a 'feedback cap' "because we can!". ;-) It is not a perfect solution, BUT it is a cheap solution. Usually, the voltage across the cap is just a few tens of millivolts, SO the cap can handle it. The reason for this is that an ALUMINUM electrolytic cap is actually 2 caps in series: one with the nominal voltage breakdown and nominal capacitance, and the other with 10-100 times more capacitance, but with perhaps only 1.5V breakdown. No matter what the DC offset is, + or - the cap can handle it. Is this good? No, better to servo, or direct couple. Tantalum caps are a separate problem, and we usually avoid them in this situation, today

Tantalum caps

Not ALL tantalum caps are bad. Just some cheap, offshore devices. In the '60's and '70's we all used tantalum coupling caps. Later, in 1978, I presented a paper at an IEEE conference on Audio, which showed significant nonlinear distortion in both tantalum and ceramic caps. Later, Walt Jung and Dick Marsh pointed out the effects of Dielectric Absorption in audio caps. So, tantalum caps can have both linear and non-linear distortion and the leads are usually magnetic as well, even with the best examples. What about a semi-defective component, made for the lowest possible price? Yes folks, you can actually measure differences between many cheap and expensive components. Today, I tend to avoid all coupling caps, and use only the best bypass caps that can be used in the price range of the product. Don't tell HK! ;-) Let them find out for themselves.

Capacitors

this is the ongoing record from my perspective. About 30 years ago, Tektronix made a modification to their 577 component analyzer in order to measure caps. This showed BIG distortion problems with typical ceramic caps. They showed this to me in Mar '74 at the factory. I modified my own 577 and ran tests. Ceramics can be AWFUL! Worse than anything else, but the TEK test procedure did not show much with anything else.

Over the years I tested caps with harmonic and IM distortion. I published this in an IEEE paper in 1978. Here, I also showed that tantalum caps (very popular at the time) had bigtime distortion when not biased with DC, as well as ceramic caps. About this time, Richard Marsh then at LBL, wrote a LTE to 'The Audio Amateur' about DA. I completely missed this potential distortion mechanism at the time, even though I had many papers on it, because it did NOT measure as non-linear distortion.
In the early '80's Walt Jung showed me a differential test procedure developed by Scott Wurcer of Analog Devices, (designer of the AD797 and many other designs) that easily measured DA in caps down to .001%. I measured 100's of caps, and found up to 6% deviation in aluminum electrolytics. Only teflon, styrene, and propylene caps were close to my residual. The rest spread out from .01%-.5% or so for film, and much more for ceramic, tantalum or aluminum caps. We published or results in 'The Audio Amateur' 4/85, and later in 'HFN'. Martin Colloms took this test and applied further measurements to caps and published several articles around this time. We have found that caps sometimes had excessive inductance, physical self resonance, and steel leads, as well as lousy contacts inside the cap itself.
Yes folks, there are differences in caps and they can be audible.
The latest work by Cyril Bateman is excellent, however he took a couple of cheap shots at Walt, me, and anyone else, who did all the preliminary work 25 years earlier. Still, he has done a lot of good work and his measurements are more sensitive, because of FFT analysis that was not cost-effective 25 years ago. I highly recommend it, with the warning that MOST ceramics are really bad, and COG or NPO ceramic caps, recommended by Cyril Bateman are the exception, rather than the rule.

I discovered using side-by-side electros for tants to reduce distortion when operating at low levels and 0 DC Volts. This was used by some designers in the early 80's. Now we AVOID tants completely. Series connection did not help for DA or non-linear distortion because the distortion mechanism is different from just reverse leakage.

Polyester is problematic. It doesn't measure non-linear distortion so much, BUT it measures significant LINEAR distortion (DA).
For the record, I used to think that polyester was OK. Then, Noel Lee, then future father of Monster cable was building one of my electronic xover designs, about 25 years ago, and he found that REMOVING the polyester caps made the sound better. I, at first, didn't believe him, BUT further work with Dick Marsh showed me that it was DA that was the problem. I immediately went over to polystyrene and polypropylene caps.

Dielectric hysteresis

Dielectric hysteresis certainly does exist in insulating materials. Cheap and popular ceramic capacitors have plenty. To accurately measure it is difficult, however. I first saw it in 1974 with a modification to a TEK 577 curve tracer, that can measure this directly. TEK invented the modification, but they did not release it as a product accessory. When you see a cap not return to its starting point, you gain understanding of this. Now, how much is in a cable? Who knows, I was measuring maybe 10% deviation with ceramic caps, and maybe cheap, high DA cable material has some too. I would not be surprised. This is the main reason that I don't recommend ceramic caps for audio applications.

Dielectric absorbtion

First of all DA used to be VERY important with ANALOG computers. Remember them? They solved differential equations, very elegantly, with mathematical modeling of integration and differentiation by using caps in different ways. A small error in the cap threw off the results. Sometimes the output went bonkers!
Heck, they modeled the polystyrene cap as a potential error generator!
Now, lets look at other caps, you know: mylar, paper, oil, ceramic. They are pretty bad, 10-10000 times worse than polystyrene.
Now, did the manufacturers of these other caps advertise their problems? NO! We had to find them ourselves. We started publishing our results 25 years ago. You might say, why did 'we' care? That's Walt Jung, Dick Marsh, and me? Well, we designed phono stages that used caps to INTEGRATE the input phono signal. We had the same problem face us as the folks did with their analog computers, in the 50s and 60's.
Now, if you think about a wire, it is basically a cap in the transverse direction, and the electromagnetic signal that flows at nearly the speed of light, is mostly in the wire covering, which is a dielectric. How good are these coverings? Heard of any quality caps being made of that material? How about that doorbell wire that you got at the hardware store? Good enough for audio?
Now, most DA as measured in caps is LINEAR DISTORTION. I know, because I measured it hundreds of times. However, normal hysteresis, to my knowledge, is NONLINEAR. Are they really the same mechanism? I doubt it, and putting them in one classification can be confusing. Enough for now.

When I first measured ceramic caps by the TEK method, 29 years ago, I found an unusual 'non-return to zero' state in cheap ceramic caps. NO OTHER cap had this obvious property. I take this to be hysteresis, because of what I saw on the scope screen. Other caps, like AL and tant can have equal or even more measured DA, which is a linear distortion and is modeled that way. DA, as measured by Scott Wurcer's differential subtraction test, seems to go away on continuous tone testing. You need an asymmetrical test signal to bring it out. Walt Jung and I wrote a paper on this 15 years ago.
Interestingly, Cyril Bateman is measuring 'non-linear' aspects of DA in some of his latest articles in 'Electronics World'. Maybe, he is on to something.
Make a differential subtractor with an ideal instrumentain op amp simulation. Cap couple both inputs, and drive them from the same signal. Have equal value R's to ground on each input. As some of you probably realize, if the caps are the same value and create the same RC time constant on both the = and - input, then they will cancel out.
Now create a model of a moderate DA capacitor by adding series RC's in parallel to one of the input caps. You can find this data by looking around on Pease's soakage article. Then try different test signals at the input, after you have slightly adjusted one side for the best null with the test signal that you are using. Try a sine wave, try a sine wave sweep, try a square wave.

Then try an asymmetrical pulse. Note the difference. Report it here. Test results should be in by the end of the week. Hint: you should find a big difference between symmetrical input signals and asymmetrical input signals.

DA is usually defined and modeled as a LINEAR DISTORTION compared to nonlinear distortion, which is measured by harmonic or IM distortion measuring instruments. DA can still be many percent of the audio signal per cap. Sine wave analysis will hide DA, so a cap with, let's say 5% DA, will have less than .0005% harmonic distortion.

First, insulation in caps, wires, etc can and do have DA. Now what do I mean by DA? I mean a tendency for the molecules to be effected by the signal in such a way that they tend to absorb some of what passes through as heat, and some as a delayed release of stored energy or perhaps electrons.
Most caps do not have a DA spec. Why? Because it is not useful to advertise it. For other reasons, they MIGHT have an ESR spec or something else that will imply the heating effects of the cap with varying frequencies. Think about a switching supply. Hi frequency, lots of harmonics, lots of cap current. What will heat up the cap? Well, lead resistance will, as well as dielectric losses of the molecules dancing along with the waveform current, etc. Some dielectrics are better for switching supplies than others.
Now what is the mystery of DA? It is normally a 'linear' distortion. This means that it will NOT generate harmonics or IM byproducts. Darn! How can you measure it then? Well, if you just charge a cap with a DC voltage, then discharge it for many hundreds of time constants (RC's), then the cap should be pretty much discharged then, right? Well, wrong most of the time. The dielectric will capture some of the input voltage and release it in its own sweet time, including several months or years. What does this mean for audio? Well. please think about it.

All that I see here is general badmouthing and half baked understanding of measured effects.
I point out at one point that you can simulate the effects of DA with a differential subtraction between two caps, using a variety of input sources, including music. Did anyone here actually TRY a simulation? If no, why not? Are you too non-technical? Do you lack the program to do it? Most likely, the most argumentative of you, just didn't bother.
And you accuse me of not following the scientific method? I know the results, as I did plots of various inputs in a differential test more than 15 years ago. I had interaction about the same results with Dr. Lipshitz and other scoffers of component differences. Even Dr. Lipshitz could find nothing wrong with my math or measurements. Today, even you 10th grade dropouts feel free to attack my measurements, yet you can't even bother to do a computer simulation, much less a real measurement.
If any of you really want to understand anything outside your past experience, you have to: Either try something, measure something, emulate something on a computer, or listen up when others, with more experience than you, give you some input about it.
The dialogue here falls short of this.

It is the TIME DISTORTION that happens bigtime with DA in caps. That is WHY you should use an asymmetrical pulse and VIEW the difference between the two caps with an oscilloscope or time domain print-out. Only the asymmetry will give you interesting results. The ratio can be perhaps 10,000 : 1.
If you do an AC analysis, rather than a transient analysis in your simulation, you will see only tiny phase differences, BUT when you do a transient analysis in the time domain, you can see up to 10% deviation in the waveform, and this is just from 1 cap! Think about 10 caps in the audio chain !!!
Does this matter? We think so.
We have known about the seeming limited effect of DA as implied by the model. This is why I ignored DA for almost 10 years. Almost 30 years ago, a fellow engineer did his masters thesis at UCB on DA. He sent me the relevant papers and his writeup on the effect, so I could have been on top of this back in 1974, when I was working with Mark Levinson and in 1977 when I designed the Symmetry 'Transient-Perfect' electronic crossover, but I thought then that mylar caps were just great! They were cheap, small, reliable, and they had no non-linear distortion that I could measure at the time, down to 0.001% IM.
Still, people did hear them, and we traced it to DA in the following years. This brought out, in our newer designs, direct coupling and servos to control offset about 20 years ago.
The person who did the first computer simulation that I know about was Scott Wurcer of Analog Devices. He had access to a good simulation program and I still have his printout. Unfortunately, I have never been successful in directly attaching anything to this website or any others, in general, so I will decline to do so at this time.

Dielectric Adsorbtion

For the record, I have done 100's of measurements on DA in caps, and this would extend to wire insulation as well, if I chose to run the tests, with wires, rather than caps. DA is a LINEAR DISTORTION so it does not show up on my measurements of NON-LINEARITY in metal wires. However, those of you who would actually like to look at some research in this area, might look at the work in 'Electronics World' over the last year and more by Cryril Bateman, regarding distortion in capacitors which goes deeper than my work of 15 and 25 years ago, respectively.

When you MODEL a component, you put in any and all potential effects that might effect performance. So, if I were to model a cap, I would put in series resistance, inductance, leakage, DA and any non-linear diode like effects to the cap model, in order to make it as complete as possible.
The IDEAL MODEL of DA does not have a non-linear component. However, it might be possible that a REAL MODEL of a cap has a non-linear component that is somehow related to DA. I don't know if this is true, but Cyril Bateman seems to think so in his articles on caps in 'Electronics World'
By the way, thanks to the individual who put Ken Kundert's paper up for everyone to look at. This should answer the most basic questions and perhaps more.
It might be noted that DA has been seriously researched for at least 50 years. It was very important when analog computers were popular, and many great papers were generated

in the '50's on DA. It was somewhat forgotten during the '60's when digital computers superseded analog computers.

At this time, I cannot exclude some subtle non-linear property of the dielectric of the wire material covering, as my reference cables both use teflon. However, I have not seen any direct correlation to this property to the dielectric used, at this time. Therefore, for safety sake, let's everyone use teflon in our interconnects
Now when it comes to non-linear effects in caps. We have measured them for decades. By far, the most interesting distortion, comes from typical ceramic caps used in cost-effective situations. This is best brought out by modifying a TEK 577 curve tracer to show gross deviations from ideal.

We don't 'demand' DB tests or 'peer review' when making our evaluations. We trust our ears, our own test equipment when appropriate, and use everything that we can to make a better audio product.
For the record, Julian Vereker and I knew each other for more than 25 years. We once hung out at each other's houses, over periods of weeks. He was located in the UK and I was in Berkeley, CA USA. I learned a lot, in the 70's, from him. I hope he felt the same way about me.

5.5 Cables and connectors

Power cords

When we want to 'win' a listening contest at, for example, CES, etc., we break in EVERYTHING, including power cords and interconnects. It is the only way that we know to get ahead of the crowd, sonically, besides bringing our best efforts to the show as well. Heck, everyone wants good sound, but attention to details makes a big difference.

I cannot prove 'anything' about power cords to your satisfaction, BUT I have listened to them with quality electrostatic headphones and have heard the difference to my satisfaction. Interestingly, a cheap thin commercial cord sounded better than an expensive model in my test, but there was a consistent difference. I also tried common mode chokes, and Bybee devices in the same set-up. I found that my STAX Lambda headphone amp is very sensitive to line cords. I learned something, you should try something too, before criticizing the rest of us.

Connectors

I could be wrong in my tests, but I have not found anything up to this time, that there is anything wrong with them.
However, I have learned something new with my latest tests. The cost of the interconnect is NOT the primary indicator of cable distortion. I have found some cheap, freebee connectors that measure almost perfectly, and the worst cable measured within the week is an expensive IC, terminated with Tiffiny connectors (original Japanese type). The

Radio Shack connectors have gotten better over the years, and only 1 out of 5 or so has any significant distortion. This was not true a few years ago, when I first tested these cables. Why? Who knows?

Cables

Electromigration also happens with copper, and it happens internally, as well as on the surface.
I point out my old reference: 'Electron Microscopy of Interfaces in
Metals and Alloys' by Forwood, starting at p314: 6.4.1 "Faulted Defects Generated by the Movement of Boundaries in Electron Microscope Specimens"
It begins with: "A striking property of high-angle grain boundaries in pure polycrystalline copper(99.999%Cu) is that they are mobile in thin-foil electron microscope specimens at room temperature and rotate during observation..."
And it goes on from there with pictures and everything.
Of course we can only see the SURFACE, because we cannot look inside the metal itself, but the mechanism doesn't have to be a surface effect, exclusively.

Cable directivity

I would like to point out that real physics of materials shows that there is much more to a strip of metal, than just its gross characteristics. Usually these subtle effects first became apparent at very low temperatures when the S/N is improved by quieting lattice vibration, but they can also be seen, in many cases at room temperature. I have noted several references on other websites in the past, and SE is aware of this.
Here is one in particular that gives me better understanding of metal properties: The book is 'ELECTRON MICROSCOPY OF INTERFACES IN METALS AND ALLOYS' CT FORWOOD, LM CLAREBROUGH ISBN 0-7503-0116-3
pp 314... "6.4.1. Faulted Defects Generated by the Movement of Boundaries in Electron Microscope Specimens
A striking property of high-angle grain boundaries of pure polycrystalline copper (99.999%Cu) is that they are moblile in thin-foil electron microscope specimens at room temperature and rotate during observation, preferentially at the surface intersections, to become more steeply inclined to the plane of the specimen surfaces. ... " (the electron microscope is turned off when not viewing and the boundries can be seen to drift with time, usually a day or more)

My associates hear differences in wire direction, and I have been there when it has been demonstrated.
It is not easy to explain just WHY this is so, but serious designers have put forth opinions on the subject over the years.
I put forth ONE EXAMPLE of how different a strip of COPPER can behave, in real time, at real world temperatures, when looked at under an electron micrcoscope. How else would we see such behavior? A small copper sample area expanded enough to include a whole wire would be perhaps 1 million times magnification, but the small sample would

still be representative of the behavior of the whole wire. Is there anyone here who can not understand this?

I attempted to ask this question: If a pure piece of copper, perhaps shaped to be conveniently looked at with an electron microscope has interesting properties, what would we see with a piece of copper wire looked at under the same conditions? How different would it be? If it is significantly different, then GEOMETRY and SCALE must be really important. However, I suspect that a piece of copper wire would look and behave essentially the same, maybe worse, if it has more impurities in it.

cable and component directivity

Just try the cable in both directions. It is easier to listen for a difference, than to see it or measure it. I believe that it can be important, sometimes.

Power cords

I can either ignore input that power cords make a difference, even in good equipment, or I accept it, when crossing every 't' is important, and make the extra effort. Of course, I must also disregard my own experience.
Actually, if analyzed, power cords do have several characteristics that people often not think about. For example, they are RF antennas! Almost the same length as our auto antenna. Also, the current waveform driving most electronic equipment is NOT anything like a 50-60Hz sine wave. What does this mean?
It is possible to experiment and find audible differences by using different configurations of line cords. Try it and see. It may not mean that one type of line cord is perfect for everything, or that it matters with your toaster, for example. Still, we have heard differences, so I have to take it into account

Cable measurement setup

This is the test:
You get some audio interconnects, they can be any length.
If they are shielded, so much the better. If they all have RCA jacks, that makes it easier to change them around, but BNC to BNC connectors will do OK as well. You can experiment with level, but 30 mV AC, 1-5KHz should be good for you. I have found that I get more effect, in general with a load on the cable. At present, I am using a 2K load, with a 600 ohm drive source. When you first make your linear measurement, without any digital processing, you will always see noise at the scope output. This is because ANY residual distortion is below the noise, both equipment or DUT. When you do a spectral analysis with the digital part of the system, then the noise will be highly reduced by the effective bandwidth of the spectrum analysis. In my case it is machine set at about 100Hz. Yours will probably be narrower.
I resort to signal averaging to get my final results (100) but you will probably need much less if any.
I look for changes in harmonic generation from the 5th to the 9th harmonic.

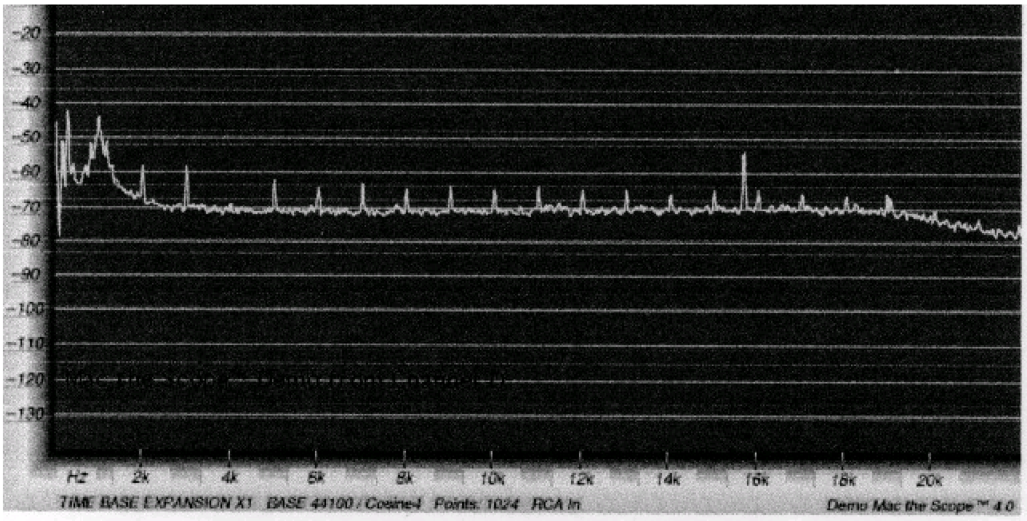
Of course, if nothing is there, no matter how deep you go by signal averaging, then that cable is free of this effect. It is best to try different cables, even cables in your lab, to see if you can measure anything.
The cost or external appearance of the interconnect does not usually give an indication of whether it will measure higher order distortion. Some very expensive and well made cables may have a great deal, others none at all. My reference cables are now a copper VDH video cable that someone made up for me to try, and the JPS cable which is essentially copper-aluminum tubing made for CATV. They measure virtually a flat line on my system residual. Also, very heavily used cables tend to measure very low distortion, all else being equal. Give it a shot.

Cable measurement setup

I admit that I am working at the 'hairy edge' of my primary test equipment, but I still get results. What is interesting to me as well, is the amount of garbage in the air above 20KHz. Now that I think about it, this might be why Bruno has so little 'garbage' in his graphs. These days, working with a 50K bandwidth FFT, I get plenty of extra stuff. Getting beyond this might be worth rebuilding my ST1700B.
By the way, I hope to get the schematics for the AP analyzers soon. I'm sure that I will learn a thing or two. Still, the shorted input noise of my modified 1700B is probably quieter than the AP, according to its specification. This is because I modified the input stage with quieter IC's

Cable distortion and "micro diodes"

Hate to be the one to tell you, BUT there are diodes in your metal wires. More than you will ever bother to measure. I have measured them.
Here are three distortion spectrum plots of John's measurements of three different interconnect cables; a Radio Shack Gold series interconnect, a model from JPS Labs, and a model from van den Hul respectively:

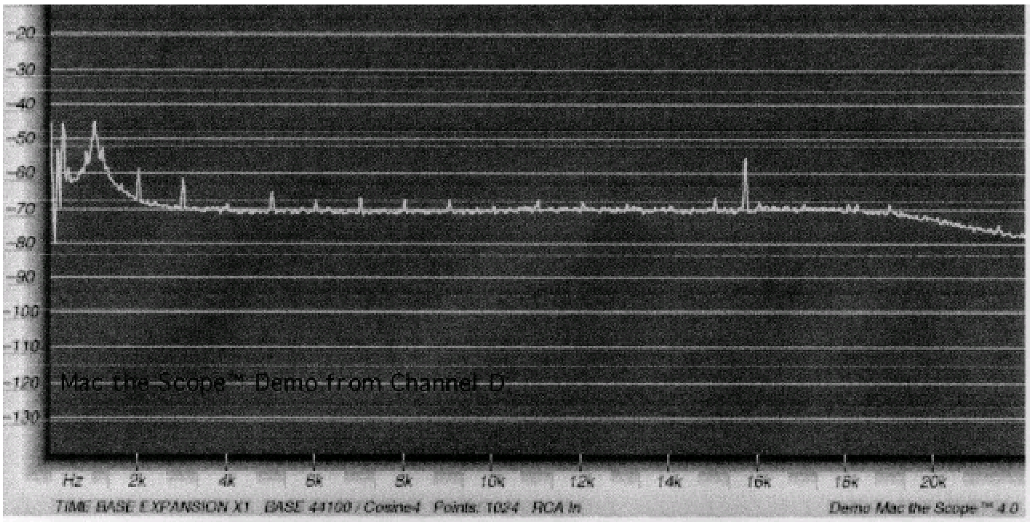


1K HZ

RADIO SHACK
(1M)

≈.012%

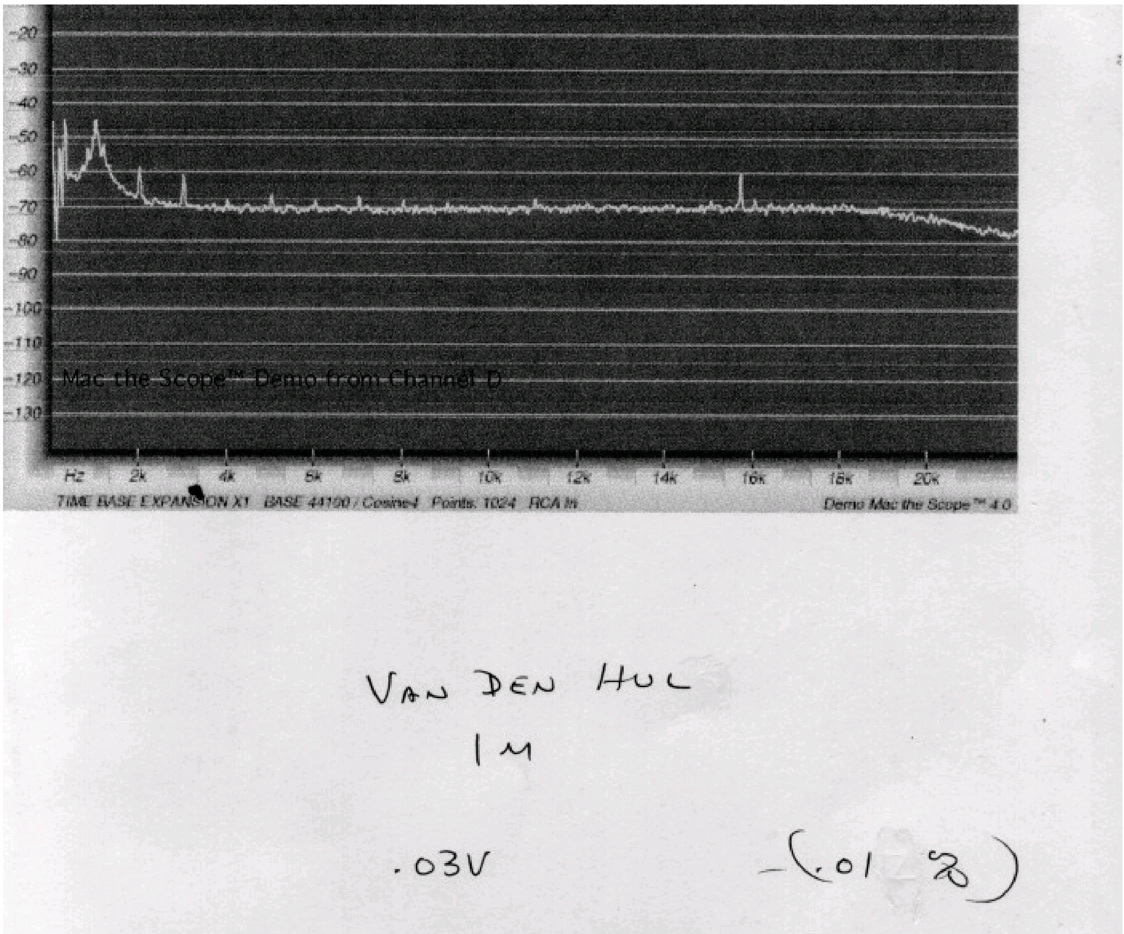
.03V



JPS

.03V

(.0122)m



I have and do measure differences in cables at levels of approximately -110dB below the test signal level, BUT I have to use lower level signals than are typically used with a THD analyzer. The graphs that I put out, that have now been put on this website, were out-takes from a series of tests that I did several years ago. They were an example of my test procedure, but they were not my best measurements. These measurements were put up in response to someone deciding that cable break-in was impossible, because he measured the cable with a Fluke multimeter and found no difference in DC measurement before and after a 'break-in' period.
I have never heard the end of it, but I still can and do make similar measurements.
Am I actually measuring cable distortion? I can't be absolutely sure, but I can say this: My measurements are repeatable, and each cable type has its own harmonic signature. Also, if I do not keep my connectors clean, I get a build up of harmonics even on my reference connectors, which will go away if I clean my connections with isopropyl alcohol. This implies some form of diodic distortion. I also know that the order of distortion changes at higher working voltage levels, like 1V or so.

For the record, the reason that the 15.7KHz interference signal is different in one of the measurements is possibly that someone in the building turned off their TV while the test was being run. This happens frequently. Also, it could be that the solid 100% shield of the JPS cable (it is like a solid aluminum tube rather than a normal wrapped shield), is filtering out more of the airborne 15.7KHz signal, compared to the other reference, or the Radio Shack cable.

Look folks, I am just trying to understand how wires work. I can measure 'differences' even today. Is it exactly 'microdiodes'? Who knows? Seems like a good hypothesis. Could it be something else, that gives similar results? I am open to any real suggestion. So far, no effective suggestions have yet been offered that are consistent with the actual test set-up.
For those who would like more info on this hypothesis, I recommend looking around the VandenHul website and look at VDH's comments.

we are now talking about 3 separate possible distortion mechanisms. First, your primary interest is diodic distortion from current moving across strands probably already oxidized by exposure to air. This could be very real. Second, we have distortion in the external contacts of a cable as they are also exposed to air and vibration (yes, this matters). Third, is the distortion that I was looking for which would be intrinsic to the wire itself, and would be there with solid core as well as stranded. I think that all three are possible, and probably measurable.

cables were directly connected to the active devices, BUT they are resistively buffered on both sides, with more resistance than would typically be used with an audio stage. The reasons should be noted, first the oscillator operates at about a 3V level and this must be resistively attenuated to 30 mV, at the same time a 600 ohm drive impedance is created. On the input, there are relays across each input to protect from overvoltage. These relays need to be current protected to keep them from being destroyed the first time they are used. Therefore, each input, both hot and ground have a 500 ohm resistor in order to limit peak currents. This has not been changed in my equipment. Therefore it is very difficult to see how a specific cable can effect my equipment to make different measurement results.

Does someone else measured distortion in wires?

I know that VDH did, years ago, but I have been out of contact with him over the years. I wish he would give me some input at this point. Frank, once you, or any one else are set up, it should be fairly easy to measure this distortion. It is just that it lies just below virtually any single function analyzer. I use two series analyzers, Cyril Bateman uses his own homemade notch filter and a pretty good A-D with a PC to get his results. The cascade has both a linear notch filter and then a FFT section that does essentially what the combination of my two units does, but they have a much better notch and a damn good oscillator. I work around this by IGNORING the notch and the first 2 harmonics (2'nd and 3'rd) These can also change but not by much. My interest is in the 5,6,7,8,9 harmonics mainly. I pretty much ignore everything else. I am still getting results, but I don't think that I have heard the last from SE on this, and I doubt that you will either. Talk about taking it personally.

Wires

I think that the microdiode model fits well within a modified varistor model. Think about ceramic grain boundaries, not with zinc, but with copper, or other metal impurities. Frank, I first suspected that you were referring to oxidization of strand surfaces which is greater in stranded wire, because of the greater surface area exposed to air, either during processing or over time. Now you hint at the wire/insulator interface. Interesting.
The reason I suspect this, is the close relationship to diode-like behavior, and the possibility of steep slopes that would easily give higher order distortion. It fits the measurements.
This is an example of how we learn new things. Not just by reading a book, or technical journal, although I have perhaps a thousand of them lying around my apartment, not just by taking a course, although I have taken more technical courses than my average critic, or just guessing.
First, you note a phemonena, maybe a difference in the sound of different wires. This is confirmed by your associates, who even may have heard it first. Then you try to measure differences in wires. Perhaps you find differences, even some that you are surprised to be there. You attempt to hypothesize the mechanism. You attempt to find the direct mechanism in the literature, if possible, and you try to fit any found mechanism into your measurement data. This might NOT be the scientific method, but it works well enough for me.

Cables

First, I measure wire and find distortion where it should not be, why? Could it be that Maxwell's equations were really derived from observing wave motion on a canal and not in a wire?
Do you think that the folks who ran the tests and observations that I cited don't know about such things? I suggest that you contact Forwood or Clarebrough at the CSIRO Division of Chemical Physics and Materials Science & Technology. You should be right at home.
Hate to be the one to tell you, BUT there are diodes in your metal wires. More than you will ever bother to measure. I have measured them.

ust because VDH and I have found distortion at low levels in metal wire, don't think that DA is not important. It is yet another problem with wires. We use teflon for our best caps and wire insulation. We have found that it makes an audible difference in our best designs.
SE is always misquoting me or putting words in my mouth. Also, wire distortion is located at about -110dB with single audio tone, in my work. I have a noise residual around -125dB. This is with 100X FFT noise averaging.

Cable measurements details

My test frequency is 5KHz and my level is 70mV with a 600ohm source and 2K paralleled with 50K load.

I am making a series of fairly difficult measurements with the test equipment that I have. My ST was designed to work to approximiately -100dB, at measurement levels about .3V or more. With added FFT processing, I can separate the harmonics from the null residue and reduce the noise floor.
I CAN produce measurement artifacts, if I am not careful. For example, if I load the oscillator buffer excessively, I can begin to produce extra 2'nd and 3'rd harmonic distortion. However, when I find a 'clear' area of measurement, then I can make measurement comparisons.
Once I have a measurement set-up that seems OK with my reference cables, I make NO changes to the measurement equipment. I just change the cable by removing it from the connector and replacing it with another.
At this point, I can see differences between different cables, both of different lengths and the same length. I can also see differences with similar cables, but with different use patterns, such as the amount of signal passed through them over time.
I can also see minor distortion 'bumps' build up, if my reference cable's connectors and the the whole external assembly are not periodically cleaned with industrial purity isopropyl alcohol The 'bumps' go down or away, after cleaning.
I monitor the input of the analyzer with a 350MHz Tek 485 scope, running maximum bandwidth, given the test conditions. I see no oscillation on the scope.
At this point I can do no more to show anything.
If Bruno's results are again a null, then the comparison is ended.

You have heard of 'dirty contacts' haven't you? Well, I seem to be able to measure them. I start with some adaptors that I must use to make the test possible. I clean them at first, BUT over time they seem to get 'dirty' I clean them again, and the extra distortion goes away. Is this an impossible concept?
You also have heard of 'break-in' of cables. It seems that when I find a particularly 'bad' cable, I tend to use it for testing more often, sometimes accidently leaving it in the machine running with a test signal for days. You might ask, am I not paying attention? Yes and no. Sometimes I turn off everything EXCEPT for my ST analyzer. It likes to be on all the time, and with something connected to it. If I forget to replace the test cable with a reference, the test signal will continue to flow through the cable until I go back, which might be days later. Just last week, I accidently left SE's steel leaded cable in the analyzer, overnight. Well, overnight was not enough to change it much, so I can still use it for testing. However, many of my original RS cables that much is still made of, have changed for the better. Darn, but I still have one RS cable that measures pretty poorly, and I have yet other examples of cables.

Passive intermodulation

I think that this input on PIM is important. I learn from it, but then I design hi end audio equipment for a living, so I don't easily dismiss distortion generation from any potential source. It is attention to details, such as whether you use a cheap audio connector that is made of magnetic materials and / or has nickel plating just under the gold plating. We have evaluated components at this level, for many years. In fact, we can change the sound of an audio component by just changing the connectors. Can't measure much however. Probably, it is because of limitations in our test methods and equipment.

Cable test setup and measurement equipment

For the record, this is the situation about my wire 'measurement'. I still measure DIFFERENCES in shielded cables with my test setup. I doubt, at this time, that it is due to distortion in the center wire, itself. This was my original hypothesis, due to the fact that Dr. Vandenhul had measured wire distortion with a different test set-up. However, on further investigation of what Vandenhul had measured, he was operating at a much lower operating level than I can get my equipment to operate. I can also see differences between clean and dirty contacts, and the presence of mumetal near the wire.
At this time, however, I don't know where the distortion is coming from, and Steve Eddy doesn't either.
I do not promote this test any further, because I have run into a 'dead end' where I can measure differences, but they do not reflect similar measurements of similar cables on other equipment. Are there diodes in wires, etc? Of course there are. Virtually any impurity or oxide should create a barrier of some kind and amount. I don't think, however, that this is the main component of what I am measuring, which is unique and repeatable for a given wire configuration.

This of course, is very important to me, because I still make and use moving coil preamps that work approximately 1000 times lower level than a typical digital or tuner input. Interestingly enough, one of his cables (the only one that I have) has always measured very well in my test, along with a few others. This led me to believe that I was measuring what he was talking about, but my working level is typically 30-70mV, and his measurement was MUCH lower, perhaps 60dB lower, and so I am not emulating his measurement.

For the record, I have a lot of test equipment gathered over the last 30 years. It is NOT brand new, but averages about 15 years behind what is available today. I have modified some older test equipment like my Sound Technology with better, lower noise, IC's for lower distortion and it was calibrated by the factory about 10 years ago, where they entirely replaced my circuit boards while they were at it.
I developed this test setup which would cost about $30,000 if it was purchased new when it was last available in order to measure my power amp designs, especially at the transition between class A and class B with a 4-8 ohm load. As my designs use negative feedback, and run fairly heavy idle current, this distortion is difficult to distinguish with a standard Sound Technology, or the HP339 which were standards in the industry for many decades, and are still useful for lab testing. As they are NOT computer controlled, they don't work as well on an assembly line.
Well, in my latest JC-1 design, this transition shows added distortion at -115dB at about 10W when I switch between high bias and low bias, which is available on the amplifier. Now many of you don't design amplifiers, so it might surprise you that I think that this distortion is important, but if I can measure it, then I can optimize the amplifier components for lowest distortion. It is like making a better running engine in an

automoble. Those of you with cheap American cars probably don't know what I am talking about ;-) but European and Japanese car owners know what I am referring to. Now, do I believe that 110-115dB down is directly hearable? No, not as a single tone, BUT as multitone IM generated with a higher order nonlinearity (kink in the transfer function), perhaps. In fact I am counting on it.
My wire tests come from developing this measurement, and unfortunately an individual connecting wire can measure -115dB down in some cases, so I am stuck with having to test my wires for testing equipment. It may not be the wires themselves, but a system interaction. I don't know yet.

I have some opinions of what some cables sound like. Some are 'bright', some are 'dirty' and still others are dull, or too smooth. I have enough cables in my lab stock to last a lifetime, IF I could believe that my opinions are imaginary. Convince me!

Tara Labs cables

I feel sorry for the people at Tara Labs. Sounds like a 'whistle blower' action just to make trouble. You know, like your ex wife turning you into the IRS, Internal Revenue, or whatever they call it in your country. Many of you here apparently have little or no understanding of what it costs to run a small business. Even if Tara bought its cable at the lowest possible price, it would still have to be terminated and this takes time and money, unless it is done like the cheap, throwaway cables. I hope the best for them, as I doubt that they did very little wrong, and as they were kind enough to send me cables after the firestorm destroyed all of my possessions.
I still have their cables in my lab. They are well made and 'measure' well, but they have a problem of being too STIFF, and ripping out connectors on the back of my TV and other cheaply made equipment. In fact, that is what is keeping me from using them today. My CTC preamp is strong enough, but my sources, Sony SACD, Fisher tube FM, Nak cassette player, etc have connectors mounted too weak to be safe from being bent or ripped out by this rather stiff cable.

5.6 Tweaks

Why is it that you people have to attack these tweaks and mods, without knowing ANYTHING about them? What is your payoff?
For the record, the Brilliant Pebbles are alleged to damp acoustic vibration. I have been linked to several websites that imply that this is for real. Have any of you actually learned ANYTHING about what you criticize?
I have known the designer for years, and he has a degree in physics, like I do, so it is easy for us to relate about this subject. What about the rest of you? Do you have the background necessary? If so, have you looked into this particular subject area? Get some facts folks, or at least try something, before trashing the component.

To me, the main point is to realize that the 'facts' that we learned in school and on the job are just 'approximations' of true reality. Just poor models of how the world works, that will be laughed at hundreds of years from now.

Where does this leave us now? Well, if you trust YOURSELF, and maybe your friends, you can try different things to see if you can or can't 'improve' your hi fi system, with them.
I, personally, have found that many, 'off the wall' tweaks have actually worked consistently for me, and I continue to use them. Try for yourself first, before laughing at it.

I might point out that 'tweaks and mods' are regularly used by the military-industrial complex, BUT they are classified if they work!

Tweaks, cryoing, etc are used in any effort to get ahead of a potential enemy. Heck, the Navy is even bringing back Cold Fusion. 'IEEE Spectrum' Sept 2004. pp22-26

5.6.1 Bybee filter

For the record, there is NO ferrite in a Bybee filter.

The Bybee devices work on a quantum level that is advanced enough that it is almost impossible for anyone to understand, even the makers of the devices. NO, Jack Bybee does NOT make the quantum devices, he buys them from a manufacturer. Jack Bybee and his family/friends have to modify the quantum devices to add the power resistor and other materials in order to make the quantum devices useful for audio applications.
What Jack Bybee states on his website is essentially what the devices do, but the exact reason for why they do it is not apparent in his description. This can lead to confusion, but it can't be helped.

Now, for a little background on Jack Bybee.
As the website probably states: Jack Bybee is 70++ old a retired physicist. In the '50's, he got at masters in physics at UC Berkeley, 6 blocks from where I live today. He had previously been a Marine officer in the Korean War, and apparently found working on military projects during the 'cold war' to his liking. Jack worked at a major company about 40 years ago in the SF Bay area in a group they called the 'Bumblebee division' at least as a joke, because the projects they worked on, 'could not possibly work'.
Over the years, Jack worked directly with Richard Feynman at Caltech as a consultant on superconductivity, and still likes to play around with physics projects. His 'quantum purifier' is one of his projects. He does it to keep busy, he is already well off and has been 'retired' for years.
I have a BA degree in physics, myself. That is not saying much, BUT it is far more that Steve Eddy has, and my conversations with Jack Bybee over the last 7+ years has re-awakened in me an interest in physics, as I had put it aside more than 35 years ago, in order to be an electronic design engineer.
When Jack and I talk, we talk physics, not money or any other BS. For example, this week I noted an article in the 'Scientific American'Nov '03 called 'All Screwed Up' on p.22 This is about an unexplored property of light.

Jack suggested that I put this up as one of our ideas, in order to get it attacked by Steve Eddy and then, later give him the reference in 'SA', but I would prefer not to give Steve Eddy any future ammunition to work with, so I just point it out here to the rest of you. For all it's worth, you can take or leave the Bybee devices, but you can't understand their operation by pulling apart what is on the Bybee website, or baiting people to give more info.

I tried the Bybee devices before I knew anything about them. They worked for me then, and they work for me now. In truth, in some situations, the Bybee devices very well could remove signal artifacts as well as other noise. So you try them and see if removing 'glare' for example, is worth perhaps some subtle signal artifact deep in the noise. This is a subjective judgement.
For example, I have found that with batteries powering my equipment, I prefer NOT to use a Bybee. However, anything plugged into the wall seems to benefit from them on the AC line. The very best hi fi playback seems to depend on the taste of the listener. Bob Crump and I generally don't like to use Bybees in the audio path of our reference preamp, or in our JC-1 power amp, because they can tend to 'lose' a small amount of 'information', but I use an inline Bybee from my video input coming from a DVD or VCR. I also use Bybees in the AC of all Video, digital and preamp inputs. We generally find that Bybees don't do much for power amps, for some reason.
The most striking place that I found the Bybees useful is in loudspeakers with bright, forward sounding tweeters, like my WATT 1's. The next most important location was in the AC line connecting to my STAX Lambda Pro headphone system.
When we go to CES, we will be using Bybee devices in our AC inputs. We have found it depends on the 'garbage' on the AC line, and interestingly, some locations don't seem to have much 'garbage' on the AC line, but CES is usually a worst case location.
I hope this gives you, more open minded individuals. an example of how Bybee devices are used and when and where they tend to work. No one expects any of you to actually invest in a Bybee. They are usually most worthwhile with very good audio systems, where a great deal of money, time and adjustment is already put forth. In these situations, the Bybee devices generate a lot of enthusiasm. In more midfi applications, they are just too expensive to consider for what they do to improve the audio or AC.

It's true that the bumblebee has recently been proven that it can fly, BUT 40 years ago, that was not the case. Therefore the name, in case anyone is confused about this.
Let's say you are in a cold war, as we were in 40 years ago and had lots of money to try things. Would you not try and use everything and anything that seemed to work?
I think about England during the 2'nd World War, and its codebreaking. We are NOW only getting some of what really happened, 60 years later. Why? Because even after the war, it was still classified.
I don't like to work at this level, and I have never worked on classified projects, and in truth, I am known to have a 'big mouth'. DuH! ;-) Therefore, I have insisted that Jack not tell me anything that might compromise him, because I'm sure it would slip out in the heat of discourse. Even today, I had to erase most of one of my messages here, once I realized that I had said too much.

This cramps my style, but I have to work within certain constraints, or else not tell you folks anything at all. I do my best with the situation. SE will tell you that I am doing something else, but I ask you, what has he done for audio that adds to the state-of-the-art. Use input transformers?

It's true that the bumblebee has recently been proven that it can fly, BUT 40 years ago, that was not the case. Therefore the name, in case anyone is confused about this.
Let's say you are in a cold war, as we were in 40 years ago and had lots of money to try things. Would you not try and use everything and anything that seemed to work?
I think about England during the 2'nd World War, and its codebreaking. We are NOW only getting some of what really happened, 60 years later. Why? Because even after the war, it was still classified.
I don't like to work at this level, and I have never worked on classified projects, and in truth, I am known to have a 'big mouth'. Therefore, I have insisted that Jack not tell me anything that might compromise him, because I'm sure it would slip out in the heat of discourse. Even today, I had to erase most of one of my messages here, once I realized that I had said too much.
This cramps my style, but I have to work within certain constraints, or else not tell you folks anything at all. I do my best with the situation.

Bybee

> Anyway, since the Bybee Quantum Purifiers are just a 0.02 ohm wirewound resistor hidden inside a ceramic tube and some heatshrink, why not just get some 0.02 ohm wirewound resistors and try those before paying $85 for a 0.02 ohm wirewound resistor hidden inside a ceramic tube and some heatshrink?

Physically, this is what you SEE! A really good metal film .02 ohm resistor, surrounded by what appears to be a ceramic form. BUT, it is not just a ceramic form. It has a layer of 'something' on its surface, AND it has silver endcaps that connect to this surface and the resistor leads. DUH! Maybe, just maybe, it is more than it looks like.

I KNOW that the Bybee device is real, maybe too subtle for most, but real, nonetheless. I have HEARD both positive and negative results with the Bybee, myself. Am I crazy? Why not always positive results? It appears to do 'something'.

Most of us are familiar with Johnson noise. This is a noise formula, that is derived from quantum mechanical equations, that is easy to apply. Basicially, it states that a 10 ohm resistor will have .4nV/rt Hz noise at room temperature, if it is a PERFECT resistor. I'm sorry that I can't make the last statement even easier to understand by those who have not studied this stuff, but I can't.
However, all all 10 ohm resistors perfect? NO! In fact, many cheaper resistors will have lots of EXCESS noise, depending whether there is an AC or DC current flowing through the 10 ohm resistor or not. This noise can completely overwhelm the intrinsic Johnson noise, and is sometimes referred to as 1/f noise.

Now this EXCESS noise is always present to some degree, and I suspect that this is the noise that the Bybee device addresses.

Bybees are real, Bybees work. Bybees are also deliberately designed to be as close to an ideal resistor as possible and to have little capacitance or inductance. This is important, and why a Bybee device will measure almost exactly like a .02 ohm resistor with conventional test equipment. However Bybee devices have been measured reducing noise with specialized test equipment.

5.6.2 Shakti Stones

Shakti Stones are microwave absorbers. There are published tests on what they absorb. There is a patent on one portion of the device. Are they useful? It most probably depends on where you place them in your audio system, AND whether there is the presence of microwave or high RFI energy in or about your audio system. They way you people carry on, one would think that they are plastic 'rocks'.
It is the same with the 'Brilliant Pebbles'. This is not a jar full of pretty rocks, any more than a good wine is just a bottle of fermented grape juice.
I read the 'white paper' on the Brilliant Pebbles and they are vibration absorbers. Now, how do I know that they are 'special'? Well, I know and personally talk to the manufacturer of these 'rocks in a jar'. Now, what is his background? Well, he is a nice guy, masters in physics in hydrodynamics, knows a lot more about mechanical vibration than I will ever know and has worked in his industry for decades, including NASA, Goodard, Lockheed, and is presently employed in Wash. DC with the FAA. It bothers him that you folks and others have no idea of what he is doing, never try it, and laugh about what you are ignorant of. What is the point?

For the record, for everyone, and back to the Shakti Stones and the Randi challenge: The 'challenge' does not hold legal water with regard to the Shakti Stones, because the device uses measurable qualities, ie reduction of RFI and microwave energy, rather than any extraordinary source to work. This is in the 'fine print' of the 'challenge'. This has been explored by legal people from Shakti. Heck, 1 Million dollars? Worth a shot. ;-)
Another misunderstanding: Shakti Stones and Brilliant Pebbles are just cute names for 2 very different products. The Brilliant Pebbles are designed to absorb mechanical vibration. The Shakti Stones are designed to reduce RFI from about 1Meg Hz into microwave frequencies. This is also why they did NOT work for my application of reducing low RFI from 5KHz to 50KHz. Aluminum foil did work, however, in that range, very well. Thanks, SY.
I hope that many of you have come to see that you were criticizing specific tweaks, and the individuals who represent them, without any real evidence.

Who knows, BUT I might try one on my new DVD player on the main processing chip. I also might put one or two in my cars, as they appear to have a measurable improvement in getting more HP out out the engine. In any case, this is a REAL device that does something that can be measured in different ways.

It just so happens that the guy from Shakti told me that several of the measurements were made with extremely advanced dynamometers, both here and in Japan. Some were VERY accurate. In autos, as well as audio, it is the small changes that can accumulate to extra-ordinary performance.

Just last night I saw a repeat TV program on NOVA that related the story of a clockmaker in the 18th century who made a VERY ACCURATE clock. In fact, it was so accurate, that no one from the educated class or nobility wanted to believe it was possible, even though they put it through YEARS of trials. After 40 years, finally, King George of England heard the story and awarded the clockmaker his due reward.
It is the same here. SY, by suggesting that Shatki stones are worthless, except as rocks, and others by demanding 'proof' without even believing, researching, or independently testing the proof put forth.

Yes, and my friend and associate, Jack Bybee worked as a consultant to the late Richard Feynman, and knew him personally. I want SY and Jack to have lunch together sometime. Might help. You never know. ;-)
Here we go again: First, someone wants measurements, then the independent measurements are discounted. Then, someone wants patents, and patents are discounted. This is a no win situation for anyone. Wake up everyone!

This million dollar thing is a set up in the case of the Shakti Stone. It has AlREADY been checked out legally. Talk to Ben at Shakti, if you want further specific information. Folks, the Shakti stones are fabricated RFI absorbers that work on a transformer principle to convert a passing through RF field into heat.
Is this so hard to understand? Is this in the domain of the 'supernatural' or 'metaphysical'? IF not, I am told that it does not meet the challange criterion put forth by the 'Randi Challange'. This was explained to me, in detail, by Ben, chief designer, at Shakti. He had a lawyer colleague look into it.
Work it out for yourselves, folks.

I just talked again with Ben Piazza of Shakti. You, who have little faith, should give him a call at: (310)459-5704 I told him already about the thread. He told me that the Dyno used on his website is a Mustang and it has perhaps a 1/10 HP rating. He has also been tested with a Dyno Dynamics made in Japan, which I think he said cost $350,000 new. It is even MORE accurate.
SY, have you read the actual LEGAL document that is associated with this test. Where does PARANORMAL come in with regards to Shakti Stones?

The Shakti stones are RF absorbers that are to be used near RF sources or potential RF recievers.

I just want to clear this up, if possible. First, Ben has 2 types of info on his website. He has his: patent, RF measurements, and even auto performance evaluations. However, he also has pictures of customer's uses of his Shakti Stone in all kinds of 'improbable' places. Let's just say that I believe the 'probable', but I personally take no position on the

'improbable' because I can't see an obvious reason for it, perhaps a real brick or stone could do just as well in many locations shown, and also I wasn't there to listen for a difference. Now, I realize that many of you JUMPED on the 'improbable' without really looking at the 'probable'.
I personally believe that the Shakti Stones work as RF absorbers, BETTER than aluminum foil, in many applications. These applications include, RF generating, or sensitive to RF, IC chips. INPUTS and OUTPUTS where RFI could be important, etc etc. I make no argument for the 'improbable', except to say that the customer can pay his money and take his choice.

unfortunately there are many engineers and scientists who have limited insight to what is possible or what can work. This includes audio, bigtime!

I spoke to Jack Bybee about this subject, a few minutes ago, (that tweaks aren't used by the military) He broke into incredulous laughter!
Also, who says that you can put a Shakti Stone or a Brilliant Pebble bottle just anywhere, and get any effect? NOT the manufacturers. You have to intelligently experiment. I was told to put Shakti Stones on my processor chips inside my DVD player. Makes sense to me.

Actually, Bybee devices have been measured, and Bybee once showed this noise reduction in a graph generated by an AC power line analyzer. The change was pretty small, but I have heard the difference in what they do, often enough. What amazes me IS when the Bybee device actually negatively effects the sound in some locations in some systems, including mine. Why?

Jack Bybee is moderately well off, he gets himself INTO TROUBLE, from his wife, for investing time and money into making audio products. He does most of it in his garage, these days. He was born in 1929, a long time ago, and he doesn't have to do anything, except for the love of audio. He also likes cars, boats, good food and wine, but audio is where he puts his extra effort, in order to improve audio systems. He could care less if someone doesn't want to try his stuff, and he offers a moneyback guarantee. What more can you expect? Double blind tests? Heck I don't even do them with my amps, anymore. Better measurements? Even I don't even have the test equipment in order to measure them properly. A clear explanation of how they probably work? It's either poorly understood or classified. There are no papers out there that give a definitive explanation as to how these devices work. Still, I use Bybee devices in my audio and video equipment. They work for me, and they work for many others, BUT I have been in situations where they DIDN'T WORK in a positive way, but actually made the sound too soft. Why? I can't be sure. Still they did SOMETHING, when they should not have made any difference, given typical measurements, any more that a few inch piece of wire.

Jack Bybee does not intentionally lie on his website, and over the years, I have tried to get him to clean it up. It was first written by his former business partner who was a CFO for Linear Technology at the time. Jack should have edited it better, from the get-go. Still, it is vague, and it will remain so. The 1/f noise is real, however, and well could be what is most easily measured. I don't know, nor does SE, whether the signal is lost with the 1/f noise. The fact that it does anything at all is a minor miracle. ;-)

5.6.3 Brilliant Pebbles

As far as the 'Brilliant Pebbles' are concerned. I know that Geoff Kait, the 'bottler' of 'Brilliant Pebbles' has been in the vibration damping business for years. It may be, or may not be, difficult to get measured info on what they do, just because of the test set-up necessary. For example, I would have a difficult time, myself, even though do I own an accelerometer.
Personally, I just don't have to have 'proof' in order to be satisfied with a tweak or a mod, or just leave it alone.

For Brilliant Pebbles, the advice is to find strong vibration nodes in a room, and put them there. I looked around with my sound level meter, and found several potential areas that could use some sort of treatment.
Neither of these devices are MAGIC! They work, based on sound engineering/ physics principles. Read the 'white papers' on these two devices before criticizing them.

5.6.4 Cap across mains

We have been using large value caps across the line (10uF) for many years. It works. Think it through. The example is compromised, because the lead length to the cap is too long and will add inductance to the parallel path. NOT GOOD! However, live and learn.

5.6.5 Bedini 'clarifier' listening test

I have personally seen the Bedini 'clarifier' used in an A-B test. I heard the difference, myself. I was surprised, but I don't doubt what I heard. Could I be confused (as usual) in an ABX test with the same Bedini device? Of course, it happens all the time.
It has been my experience to try things, and if you can hear a difference, then that is provisionally adequate to consider including this new component, or adjustment, in order to make the best sounding audio system. This is what makes winning audio systems, rather than also-rans, that sound OK, perhaps, but so does so much mid-fi.

5.6.6 Life energy plus

http://www.diamondcenter.net/digitalstress.html

I met Dr. Diamond at an AES convention. He is an interesting, 'new-age' guy. I read his book. I had him over to my place for a day or two. I also built some recording equipment for him.
This is what I know: Like many tweakers, Dr. Diamond is short on conventional 'proof' , but he makes sense to me. Dr Diamond is a psychologist, or something similar. His therapy has included listening to music. He did this for years. Unfortunately, when digital audio came out, he found that the positive effects of music therapy was lost. He investigated this problem and decided to give a talk about this at the AES. I was there, when he gave his paper. He almost caused a riot! It was great! Made my day.
After the sesson, Doug Sax, of Sheffield Records, and I went for a drink, to celebrate this talk.
When he came to both my lab and my home, he tested me with a number of things: For example, I then wore an analog watch, BUT I was found insensitive to wearing a digital watch. I WAS sensitive to a sugar cube dissolved in my mouth. IF I could have beat him in his testing, I would have. I don't like being fooled any more than anyone else.
However, my significant other, Karen Richardson, normally wore a digital watch, and she was found sensitive to it. We tried many things, including digital. It was very interesting.
What did he get out of it? Nothing, but some interactive feedback by a reasonable skeptic.
Does he have a tendency to exaggerate? Yes, but no more than many others in the audio world.
This is my take. At least I know the guy.

6 Troubleshooting

Amplifier troubleshooting

When first testing, USE A LIGHTBULB in series with the AC line. No kidding! This is a life saver. You will have to make up your own lightbulb socket, but make sure that it is in

SERIES of the amp. This is a very old technicians' trick. Think about it: A 100-250W 120 volt bulb will have a very low COLD resistance. If the needed idle current is reasonably low, the lightbulb will stay relatively off and still have a low resistance. However, if the amp requires lots of idle current (a dangerous condition) then the bulb will heat up and become high resistance, limiting the absolute current through the whole amp. It makes the amp last much longer, before breaking and usually you can even determine what the problem is, while still running the amp (sort of). I hope that this helps.

For the record, I also use a variac with the lightbulb, BUT the lightbulb is the most important, and many of my new designs won't even come on until a minimum voltage is reached, so the variac is not as useful as it used to be for me.

Distortion Analyzer

Do you have the time and money to build a first class distortion analyzer? Why not try to get a pretty good one on e-bay or somewhere else and modify it, if necessary? Trust me, it will save you both time and money, to get something that already is in a box, ground loops worked out, and has a good meter. If you look around, (and are lucky) you can get a good distortion measuring instrument for 5%-10% of new cost. The used equipment companies seem to charge very high prices for the same thing that you can find elsewhere. Just look around.

Scope

One very good buy, in the USA, is a TEK 465.

Scope preamp

I am currently using one of Scott Wurcer's IN-AMP designs, the AD524. It is not as quiet as the AD797, BUT it has built in gain settings, up to 1000, (I use 100) and it has a true differential input. Run it by Scott. I use 9V batteries to power it

6 Library, edication and books

Education

Do we just think it through, from the knowledge we have gained by taking a course or reading a book? Or do we try things, try to put aside our initial opinion, and if we find that something that we try works for us, we might find out how it works for others, then if it also seems to work for others of like mind and interest in improving things, we might also try to find a physical reason for why something works. At least we might make a hypothesis as to why, and change it, if we find another direction that fits the situation better. This is how I learn new concepts. It makes me a successful audio designer.
Now what about 'education'. I have been accused as an 'education snob'. Let's get real, I come from a working class background, and I only first attended college to become a TV repair technician. Once in college, a whole new world was opened up to me. I went from

tech ed., to engineering, to physics, and in real employment, back to electronics engineering. It took me years of after-hours classes to catch up and formally become a senior electronics engineer. I am not an academic, but I both learn from and teach academics. If I need a Ph.D., I hire one. Sometimes they hire me. This is the pursuit of understanding, without self limiting. Works for me.

Education

It is good to go to the university and get a solid technical education.
With a good technical education, it is possible to understand the derivation of many distortion sources, many of which are important in audio design.
There are many technical books that are useful to understanding circuit design, provided that you are interested in the subject, and don't attack every opinion offered in the technical book, UNLESS you are prepared, with a good technical education and the mathematics, to back up your difference of opinion.
If you have a good technical background, then it is EASY to learn about specific distortion types, such as TIM or PIM, by doing a 'Google' search and reading what comes from it. Some of it, at least, will be at a level that you can grasp, even if you don't have heavy math skills.
As an engineer, I am normally interested in 'engineering'. This means that I use previously derived formulas in most cases to estimate my design and its performance. I also measure what I make and see if it matches my estimate.
Deriving first principles, doing heavy math, or re-inventing what is already in a book, is not what I am interested in doing.
SO, I tend to believe the formulas and equations that I can get from a book, unless I have a good reason NOT to do so.
Audio design is based on what the ear hears, ultimately, not what you want it to do, or what someone tells you is unimportant. It is the proof of performance that counts, and why I am well known in audio design.

Education

What I learned in college was the math that I would never learn on my own. While it has been over 40 years for me since I took 'Differential Equations' or 'Advanced Calculus', two required courses in my junior year in physics, the basic concepts remain, especially when I read research material.
For example, almost 30 years ago, I 'INSTANTLY' understood what Herr Manger was doing with his exotic loudspeaker, once he gave me a reprint of the 1925 article of 'Rice and Kellog' on the design of the loudspeaker. The differential equations with dot notation form were readable on inspection.
Without my 'required' math background, I could still be scratching my head over what is really unique about his speaker design.
This is the power of a 'strong' technical background. You have to learn the tools that people, over 100's of years, have developed.

Now, did I find that getting through college actually 'educated' me? Not when I actually worked in electronic design. At first, I felt completely helpless, but with the help of a few experienced engineers who I worked for, and literally 100's of lunch hours in the company library, I was able to begin to grasp electronic design, and do independent work.

A good avenue of getting exotic technical info is the university engineering library. While, I am not a student at UC Berkeley, and I haven't attended in any way or more than 30 years, I can still go to the engineering library, go to the periodicals, and copy virtually anything that I want. Almost everything in electronic engineering is there, including 'Electronics World'. There must be many situations like this around the world. Check out your own university technical library.

Now, when it comes to professors: Let me relate a story that began 1/3 century ago. In 1971, I decided to audit (sit in) classes at UC Berkeley. One of my professors who taught linear design was Dr. RG Meyer. I learned plenty from his course, including distortion analysis, and problems with negative feedback. Surprisingly, he is actually slightly YOUNGER than I am, so the professor-student relationship is a little odd.
In 1973, while attending a similar graduate course on the same subject, I visited Dr. Meyer during his office hours. He had impressed me with an analysis of 3'rd harmonic suppression and expansion with local feedback. I asked him whether a similar analysis had been done for 5th or higher odd harmonics. He said no, and politely implied that he had no further time at the moment for a student like me. This was normal, between a professor and a student.
However, in 1976, I worked with Matti Otala on a TIM measurement paper, which we presented at the AES in fall, 1976. Dr. Meyer apparently read this paper and recognized my name.
Then in 1978, I attended an ISSCC conference, here in San Francisco. There I again met Dr. Meyer and he recognized me. This time he treated me as a peer, not as a student. I had finally gained some recognition on his level.
Finally, in 1980, I became a paid 'practical' low noise consultant for a project Dr. Meyer was working on, even gave him part numbers and topology suggestions for an ultra-low noise design. This process took almost 10 years!
See how things can progress? But you must cope with your position in life until you become a serious contributor. Then, you too will find life much easier for you, and you too, will be respected by your professors.
Please, don't think that 'you' have all knowledge and understanding enough to shape your own course in your education. Every student feels like they are learning useless nonsense, BUT LATER, you find why the professors made you suffer through it. Don't give up, even though you might need a vacation from the nonsense that you appear to be suffering. Just remember, without an education, you just can't do the most interesting stuff, and putting up with the process of getting a college education is part of the test of your character.

Books

Yes, Bob Pease's book is very good. Also, consider 'Analog Circuit Design' and 'The Art and Science of Analog Circuit Design' both edited by Jim Williams of Linear Technology. Jim is a really capable analog designer.
A third significant book, in my opinion is: 'Intuitive IC Electronics' by Thomas M Frederiksen. This book gives a good insight into solid state internal behavior, or why transistors work, without the really heavy math that is usually involved.

If some of you folks would just buy 'The Art of Electronics' you would be WAY AHEAD of the game. 90% of everything that an amateur would need is in this one book. Also, it is readable. More so than many other books named in the survey. I have hundreds of textbooks, but I don't recommend them all to you, because they are specialized, arcane, and seemed interesting to me at the time of purchase. How about Maxwell's original papers? Heaviside anyone? Steinmetz? And you people talk about first principles!
The A of E book got 4 1/2 STARS out of 5! It is IMPOSSIBLE to please everybody. I know, from experience.

Books

Those books listed above look great for someone like me, who designs amps for a living, but they are most probably above the level brought forth here on this website. 'The Art of Electronics' is a pretty good general text on many aspects of electronic engineering. I have never seen a really good book on amp design, the closest being Ben Duncan' s book on amplifier design.

I have never found that KNOWING how transistors work is very important. It is how they BEHAVE, in a circuit, that is most important. The physics of a transistor is interesting, but like knowing how the inside of of human body works, not important in working with people. I STILL don't really know how a transistor works. Not well enough to design one.

The books that I tend to refer to are based on classes that I took in EE. For example: 'Analysis and Design of Analog Integrated Circuits' by Grey/Meyer, 'Electronic Engineering' Alley/Atwood, and 'Analog Integrated Circuits for Communication' by Pederson/Mayaram. While two of these books have IC design primarily in mind, I learned a lot from the courses from which these books were derived both at the undergraduate and graduate level, that extends to all audio design.

Will I write a book?

I keep TRYING to tell you how to make better amplifiers
When it comes to schematics, you have my past efforts, as well as those of Nelson Pass and Charles Hansen on this website. These are examples of circuits that work! You can't have access to our LATEST circuits, because Charles, Nelson and I are actually in competition with each other in the marketplace.

When it comes to exact design, WELL, experience seems rather important, and some educational background as well.
I guess that it would be nice if I wrote a book or even a white paper showing you how to make a 'successful' amplifier. It would be like someone showing you how to build your own sports car. Many of you would take short-cuts, use cheap parts, substitute with impunity, and then WONDER why your results were not as good as the original design. After all, are resistors really different, caps, wire? According to many here, NO! What about circuit layout, the amount of negative feedback, and the presence of higher order distortion in the circuit? Who cares?
Sorry folks, I would like to write a book, but no one has made me an offer that I can't refuse.

Library

I have hundreds of tech books at my side. How can you hope to understand, if you won't invest in a textbook, once in a while? I pulled two texbooks that were within arms reach with the info. Unfortunately,'The Art of Electronics' did NOT have any info. However, you can also go on the internet and get the same info, I'm pretty sure. I can't write the equations here. I am just not equipped.

Not enough books! However, you CAN go on the internet and get essentially the same equations. It is not as easy as some inputs, BECAUSE THD is NOT what you want, you need to know the change of magnitude of individual harmonic distortion components with changes of level. I am pretty sure the 'Radiotron Designers Handbook' has these equations as well.

Bode book

I show how a linear design engineer has to look at OPEN LOOP DESIGN, and mostly all that I get is 'static'. Bode doesn’t count for much in open loop design. OF COURSE, if we use GLOBAL NEGATIVE FEEDBACK, we can reduce our distortion to almost anything that we want. However, negative feedback has problems that even Bode can't fix. This is why many hi end designers strive to reduce or eliminate global negative feedback when possible.

I have been using negative feedback as a designer for more than 35 years, BUT it has serious problems in audio design. That is why Dr. Otala, and many others have written about its problems and limitations. You simply have not looked that far or listened to enough, if you think that negative feedback is the answer.
Basically, if I could make a high feedback circuit sound as good as a low feedback circuit, I would do it.

Books

Yes, Bob Pease's book is very good. Also, consider 'Analog Circuit Design' and 'The Art and Science of Analog Circuit Design' both edited by Jim Williams of Linear Technology. Jim is a really capable analog designer.
A third significant book, in my opinion is: 'Intuitive IC Electronics' by Thomas M Frederiksen. This book gives a good insight into solid state internal behavior, or why transistors work, without the really heavy math that is usually involved.

If some of you folks would just buy 'The Art of Electronics' you would be WAY AHEAD of the game. 90% of everything that an amateur would need is in this one book. Also, it is readable. More so than many other books named in the survey. I have hundreds of textbooks, but I don't recommend them all to you, because they are specialized, arcane, and seemed interesting to me at the time of purchase. How about Maxwell's original papers? Heaviside anyone? Steinmetz? And you people talk about first principles!
The A of E book got 4 1/2 STARS out of 5! It is IMPOSSIBLE to please everybody. I know, from experience.

Books

A few more comments: "The Art of Electronics" is one of the best general/ beginning books in linear design. I have it here, and gave one to my business partner, Bob Crump (degree in psychology), for his birthday. If someday I am not available to answer a technical question, he can refer to his book. I, especially thought that the section on low noise design was very advanced, even today.
I used to have #18 'Vacuum tube amplifiers', before the firestorm. I might just try to find it again. I remember learning about the 'White follower' in that edition.
Another book that I might recommend for beginners especially:

The 'Active Filter Cookbook' by Don Lancaster Sams, isbn 0-672-21168-8
from 1975, or even earlier. It saved my tail almost 30 years ago.

Cordell, Otala, and Gilbert papers

I think that you will find that we are going back to higher open loop bandwidth. It is true, that you do not, necessarily, need a very high open loop bandwidth to remove TIM, but you do need it to remove FM distortion.
Trust me, high open loop bandwidth is preferred, all else being equal.

Books

Those books listed above look great for someone like me, who designs amps for a living, but they are most probably above the level brought forth here on this website. 'The Art of Electronics' is a pretty good general text on many aspects of electronic engineering. I have never seen a really good book on amp design, the closest being Ben Duncan' s book on amplifier design.

I have never found that KNOWING how transistors work is very important. It is how they BEHAVE, in a circuit, that is most important. The physics of a transistor is interesting, but like knowing how the inside of of human body works, not important in working with people. I STILL don't really know how a transistor works. Not well enough to design one.

The books that I tend to refer to are based on classes that I took in EE. For example: 'Analysis and Design of Analog Integrated Circuits' by Grey/Meyer, 'Electronic Engineering' Alley/Atwood, and 'Analog Integrated Circuits for Communication' by Pederson/Mayaram. While two of these books have IC design primarily in mind, I learned a lot from the courses from which these books were derived both at the undergraduate and graduate level, that extends to all audio design.

Epilog

Actually, there are a great number of parallels from all over the word as to how to make a great audio circuit. I never cease to be amazed, for example, how many people have discovered Roderstein resistors and their sound quality in audio products. Yes, they are costly, maybe 5-10 cents each in reasonable quantity. Too high for mid fi.
As far as differences in designers are concerned, those who make the good stuff, usually share many factors, such as class A, if possible, that tubes work darn good, etc. Where we differ is what makes a 'horse race'. If everything were exactly the same, we would not be in business. If for example, there was only one standard circuit, it would be unproductive for us to build it separately, it would be given to some mid fi mass producer.
I have designed at every level. I once designed the amp replacement for the NAB approved 'All American Five' kitchen radio. Cost was $1.00 for parts. Worked pretty good too!

I want to make this clear:
I make audio equipment that most accurately reproduces recorded music as best that I can do at EVERY price point. I NEVER add distortion or allow extra distortion to remain for some subjective reason, nor do I allow my associates to do so with my designs. I always try to make the best out of what I have available.
The difference between many others in mid-fi and myself, is that I put the money were it counts, such as in Roderstein resistors, rather than 2 for a penny devices that look similar, if and when I can. With Parasound, I often have a limited role with the lower priced components, but I often can chose the IC's and eliminate the ceramic caps from the circuit. This puts me ahead of the pack, already.
I am NOT making musical instruments, but music conveyers that get you as close to the original performance as possible. This precludes making a component 'musical' in some way, on purpose.

It is NOT how much distortion that you have (within reason) BUT how complex the distortion is. Complicated circuits (and complex feedback) usually make complex distortion, composed of a long string of higher order components as well as FM modulation. Simple circuits, designed properly, usually just have lower order distortion, and even though it might be easily measurable, it will either not be really audible or perhaps slightly different from the original in an easy to live with way.

Early on in my career, I worked with or initially developed many of the push pull drive stages used today. The reason for a push pull drive stage is to lower distortion due to symmetrical drive (about 21/2 times) and to create an intrinsically stable circuit irrespective of dynamic effects. It also makes the slew rate symmetrical as well.
The example given of a class AB output stage is primitive, and a real output stage is not really as bad as all that, WITH GOOD DESIGN. Higher order distortion can be reduced to almost nothing, with some design effort. Then, the driver stage distortion can be more important, at least to me.

I still think that you confuse series 'complexity' with parallel push-pull design. Parallel push-pull design reduces the current change necessary for changing voltage and tends to cancel much of the inherent distortion as well.
Series design adds devices in series, which can reduce total distortion, BUT usually multiplies the order of the distortion as the distortion in one stage is passed through a following stage.
Both design approaches take more parts than absolutely necessary to make a primitive amp. I prefer the parallel approach. It still has a SIMPLE through-path, or transfer function.

I agree that we have gotten off the original subject. I also agree that reading Feynman is difficult. I have most of his books, but I have trouble also. I talked this over with my friend, Jack Bybee, who used to work with Richard Feynman. He told me that Feynman tended to skip over important logical steps, as if they were obvious. I found this also with the late Richard Heyser. You know, the words are simple, BUT how did he come to that conclusion? Therefore, Feynman puts me to sleep (really) while reading.
Back to the audio reference books: There are books that give one person's point of view that contain good material, such as Doug Self's book. However, it would be almost impossible to design anything different that what Doug Self designs with only his book as reference. Most serious reference books have moved away from discrete design into IC's, OP AMP theory, or CMOS. There are older books that have important chapters, such as the 'Radiotron Designers Handbook', or 'Electronics and Radio Engineering' by Terman. These old, but relatively complete texts give numerous insights to audio design if you look through them.
Finally, I would find 'OLD COLONY' on the internet and look that their audio book selection. If it is available, they usually have it, and their selection is very good.

When designing audio amps, the most important thing is to remember what you are trying to design---A GOOD SOUNDING AMPLIFIER. There are guidelines such as 'low higher order odd harmonic distortion' 'fairly high slew rate' and 'the first watt is the most important'. If you follow these guidelines, you can not often go wrong, YET these guidelines are often ignored.

Detailed differences come with experience: For example, should we stabilize a feedback controlled power amp with an output coil? We used to use them, but you will find them missing with most hi end designs,today. Why?
Should we use a push-pull class AB design, or a single ended class A design? Often the class A design will sound better, BUT remember it MUST be class A, and the push-pull design COULD be class A as well, BUT you would lose much of its potential output power, or have to increase the heatsink considerably.
IF you make the push-pull power amp class A, with the same quiescent current as the single ended design, then other design details may dominate, such as stage complication, amount of negative feedback, etc, rather than single ended vs push-pull.
However, on the other side: Should we put balance pots in our push-pull amps to null out the last bit of 2'nd harmonic, or should we leave the residual 2'nd harmonic distortion if it isn't too high? Balance pots tend to complicate the design, is this a worthwhile tradeoff?
These are examples of the decisions that we designers have to make. We don't always agree, and that makes for different products in the audio marketplace.