Subject: SPeaker driver materials: do they matter?
Posted by akhilesh on Fri, 17 Dec 2004 17:00:20 GMT
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Yesterday I heard the infinity intermezzo 4.1t system at ultimate electronics. IT is said to be very flat, very free of distortion. It did absolutely nothing for me. Of course, this could have been the electronics, but i doubt that, since most electronics are flat & distortion free. Next, I heard the $500 boom box system from JVC, that uses a single driver WOOD speaker (yes real wood) with a biflex diaphragm. I thought that sounded MUCH better. Then I went home and heard my system this morning. SO much preferable to the infinity. My system has MUCH more distortion & is probably not flat at all. THis got me thinking: just as one cannot make a good violin or a cello from a bad materials (meanng it's not just flatness or distortion, otherwise anyone could make two cellos sound tha same) maybe it's the speaker matieral, and the way it is put together that makes one driver sound different from the other. I persoanlly like wood & paper drivers. THe inifinity system used a metal alloy driver called CMMD. What do others think?-akihlesh

Subject: Re: SPeaker driver materials: do they matter?
Posted by Wayne Parham on Sat, 18 Dec 2004 06:28:37 GMT
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I don't much care for metal or plastic woofers or mids.  But I do like metal tweeters.  Cone breakup behavior is different because the stiffness of the materials is different.

Subject: Re: SPeaker driver materials: do they matter?
Posted by Manualblock on Sat, 18 Dec 2004 16:34:56 GMT
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There is an old urban myth that Western Electric used to manufacture the paper & fibre cones for their woofers by the batch, mixing the components along with the binding agents in a washing machine. Of course there was a secret formulae involved. AK the distortion thing; it has to logically be accepted that we are not measureing everything that impacts reproduced sound. There is absolutley no other explanation possible for why examples of electronics that measure the same sound different. Until that FACT is accepted you just go around in circles like a dog chasing his tail.The sad thing is; if you have to prove that distortion measurements tell the whole story, you are stuck in the water because then by definition you can never admitt you hear differing sound from one low distortion piece to the next. Consequently you are choked off from ever finding real reasons for why things sound different because according to the measurement they cannot sound different. That was the ultimate connundrum that ruined Julien Hirsch's reputation, he just backed himself into that measurement corner and watched the audio world pass him by.

Subject: Accuracy versus subjective preference
Posted by akhilesh on Sun, 19 Dec 2004 15:27:05 GMT
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HI John,I agree. I think accuracy by definition HAS to be measured. Subjective preference on the other hand has nothing to do with accuracy. For example, some people LIKe more bass than others and EQ their systems accordingly. Does that mean it;s more accurate? Of course not. I personally LIKE the midrange a bit bumped up. TDoes that mean my system is more accurate? Probably WAYY less accurate than most systems. But it sounds a HECK of a lot better to me than, for example, the very accurate infinity 4.1t speaker I heard, with decent electronics. Of course, I heard arecording I have never heard (Seal) and that may havebeen dead, asmay have been te room (which I thought was too live and bog). But the system did absolutely noting for me: the sound was not rich & meaty the way I leike it. So, yeah, I think we all agree: accuracy has nothing to do with preference. Same  example with a violin: a stradivarius may or may not measure better than a violin made of aluminum: i am not sure. But it sounds a heck of a lot better.
-akhilesh

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Subject: Re: Accuracy versus subjective preference
Posted by Manualblock on Sun, 19 Dec 2004 17:30:08 GMT
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AK; the violin example makes my point. I am not reffering to what we prefer or what we like, I refer to the concept many people have that if two amps measure the same, with the same exact results, then they must sound the same. Many people think that if an amplifier shows very low distortion; then it is reproducing the signal exactly as the signal was recorded, and that means that the amp adds no sound of it's own to the music. I have heard amps that have extremely low distortion figures compared to each other in order to see if that is true. The amps had different sound. Can you explain that?Happy holidays BTW to you and your's. J.R.

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Subject: Re: Accuracy versus subjective preference, yes
Posted by tomservo on Sun, 19 Dec 2004 19:49:24 GMT
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If one presumes that the speakers job is to reproduce the music as opposed to be part of its production (as in a guitar amp etc), then one can look at it from the perspective of the speaker deviating from an ideal device where what goes is in = what comes out.Also, fwiw, while the THD figure may be identical  on two totally different sounding amplifiers, they will still be very different if looked at a more appropriate way.It was after all the marketing of solid state electronics which caused the use of distortion measurement’s to be "an indicator" (as a marketing tool ) of "goodness" in the market place.In the case above, one would typically find one amp (the good sounding one) having higher levels of mostly 2nd and 3rd (low order) harmonics vs the "bad" sounding one having more orders but all lower level. The issue is obviously THD % does not track audibility but it is an easy thing for the sales staff to point to as a sales tool.The ideal "perfect radiator" allows the driver to act as if only the motor and suspension were acting on it. In that case (ignoring the low cutoff), one finds flat response up to the point the inductive roll off is reached and the it rolls off at 6 dB/oct. Also, if one examines energy vs time, one finds the distribution is also that set by the motor etc.In real life, one finds the radiator is much "more" than what is ideal.For one, a real driver is often operated through the range where it is an acoustically small point source into the range where it has some directivity.This is somewhat problematic as the driver’s directivity causes a shift in the acoustic phase which is not exhibited in its amplitude, that is it is Non-minimum phase and is a shift in time of origin.In order to make the on axis response flat in the face of increasing directivity (with increasing frequency), it is necessary to roll of the input power through proper choice of Le and internal absorption in the radiator.In the "directivity" range (the frequency range where the driver is larger than about 1/3 wl in diameter) the velocity of the force propagating in the material must be kept low, ideally keeping pace with the acoustic pressure from the center.The higher into the directivity range the driver is used, the more critical and eventually show stopping these factors become.Fold into this the factor that as soon as the driver departs significantly from "piston operation" it is in semi controlled breakup and it is a tough design challenge in every way.  At the top end, the modes become so complex and chaotic that the response is a series of spikes and then rolls off as the radiator is nearly fully decoupled from the motor.It is the fact that all of these cannot truly be balanced in a driver much larger than a headphone and you see why multi-way speakers are the norm.A hidden problem in breakup in general is that if one has a peak in the raw response, that peak represents a frequency selective "acoustic gain".  Some times the peaks can be large and while the way of dealing with them is to place the brake up above low pass crossover crossover, this does not really fix all of the problem.Lets say one has some big break up peaks around 3KHz in the woofers response.You use a steep crossover at say 1.5KHz, problem gone right?When operating the driver at sub-multiples of 3KHz, the acoustic gain is still present, it amplifies the distortion product at that frequency by that amount.For example, at 1KHz (in band, well below crossover) , the 3rd harmonic is amplified by the amount of that peak at 3KHz. In the systems range there is going to be a frequency where each harmonic, 2nd, 3rd, 4th 5th and so on is amplified by the magnitude of that peak at 3KHz.This is yet another way the cone material can significantly alter the "sound" or flavor of a driver without having different "in band" response measurements.The bottom line might be that the best radiator has the best trade off of stiffness, sound velocity, damping and of course cost.All of this is why radiators can be made out of paper made in a washing machine with "secret ingredients, possibly from the cat box", wood soaked in sake, plastic, metal exotic or just plane Jane pulp.None are "ideal" in every way, over the entire frequency range, all have significant trade offs, especially the wider the frequency range is.So complex are all of these factors that even with modern computing power, the best examples of great full range drivers were made by mostly by trial and error by "driver engineer artists".Hope this helps a tad.Tom Danley

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Subject: Re: Accuracy versus subjective preference, yes
Posted by Manualblock on Mon, 20 Dec 2004 02:14:58 GMT
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Which is why good speakers are rare and are voiced by talented designers who approach the art of speaker design the same way good Luthiers approach the art of stringed instrument design. Using their ear as the final arbiter of quality. Very good post. Thanks J.R.

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Subject: Re: Accuracy versus subjective preference, yes
Posted by Earl Geddes on Thu, 27 Jan 2005 00:46:58 GMT
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TomAs usual you did a good job of explaining some things, but side stepping others.For instance, just exactly what should the idrectivity of a source be?  This was not discussed.  Maybe the collapsing directivity is a good thing? in some cases.  I use this natural collapsing power/directivity response in my favor.  By selecting the right diaphragm radius one can match the directivity of the LF unit to the HF unit yielding a very wide frequency range of constant coverage (CD-constant directivity, which also means constant power), below which the coverage widens smoothly resulting in an increase in the power response at LF.  This nicely balances the increase in absorption at LF that all good acoutic rooms must have.  SInce room power is power in minus power out these factor can all be balanced such that the room sound power is relatively constant with frequency.So while I agree with your analysis, what you describe as problems can sometimes be used to advantage.I completely agree with the cone break-up problem even when it is above the X-over.  Even with a 3rd order LP, I have trouble with the inherent breakup peaks poking through the total power and pressure response.  Its the one problem that I am try to resolve at this point.  I want a 15" that is flat, or with a slight rise, to about 1 kHz, but then I want it to die - as fast as possible.  Seems that to get to 1 kHz they all have a peak just above this point.At any rate, there are many ways to skin the loudspeaker design problem - they all have their pro's and cons.

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