John Dunlavy's posts to rec.audio.* during 1999:
New messages will be added here as I see them on Usenet; hopefully in less than a week. Thanks to Dunlavy for this information.

Since July, Mr. Dunlavy has been posting with much greater frequency and in a more interactive manner than previously; I am archiving primarily the posts where he explains something I believe will be of general, enduring interest.

Last revised 12 November 1999.

Subject: Re: Sweet Spot

From recent posts I have read, there seems to be a lot of confusion regarding the subject of a loudspeaker's "sweet-spot" and the properties of a loudspeaker that contribute to the width of the listening area over which an accurate stereo soundstage can be perceived.

To begin, the term "sweet-spot" applies to two separate properties of a loudspeaker: 1) the listening area over which an accurate spectral balance is reproduced, and 2) the listening area over which accurate imaging and an accurate stereo soundstage are reproduced.

These two kinds of sweet-spots may or may not co-exist, depending upon the following loudspeaker properties: 1) the horizontal width of the listening area at a given distance over which the spectral balance (frequency response) of the loudspeaker is preserved, and 2) the degree to which the loudspeakers are "pair-matched" with respect to their frequency and phase responses, both on and off axis.

Thus, a pair of loudspeakers may exhibit a reasonably good spectral balance on-axis and off-axis and yet blur the reproduction of an accurately recorded soundstage, with respect to precise imaging of discrete sound sources.

Indeed, a pair of loudspeakers with a wide horizontal dispersion pattern may be exhibit a subjectively smooth spectral balance over a wide horizontal listening window, and yet be incapable of providing "pin-point imaging" of accurately recorded sounds.

Likewise, a pair of loudspeakers with nearly perfectly matched on-axis amplitude and phase responses vs. frequency, may exhibit a relatively narrow listening window with respect to off-axis spectral-balance.

Seem contradictory? Hardly!

Our hearing process determines spectral balance and direction (angle of arrival) of a wide-band sound source (music, for example) by different means. The sensitivity of the human hearing process to the "spectral balance" of a loudspeaker is relatively poor. However, relatively small differences in the phase response between two loudspeakers comprising a stereo pair, can markedly blur the reproduction of an accurately recorded soundstage or shift the perceived direction of a given sound at a given frequency. Thus, truly accurate reproduction of the recorded soundstage is dependent upon accurately matching the amplitude response, phase response and directivity patterns of a pair of stereo loudspeakers and ensuring that the listener is located precisely equidistant between them.

This explains why a pair of truly accurate, well-matched loudspeakers require that more attention be given to choosing the correct listening position, equidistant from both speakers, so as to obtain the most precise stereo imaging. A pair of loudspeakers with less "pair matching" with respect to amplitude and phase Vs frequency, will yield a wider listening area over which, however, the stereo image will be less precise and more amorphous.

Anyone wish to comment?

Best of listening,
John Dunlavy

Subject: Re: Electrostatic Loudspeakers, information requested

ESL's, and membrane type loudspeakers in general, present numerous demanding challenges for anyone undertaking their design. This would especially apply to anyone without reasonable competency in the disciplines of electronics, physics and acoustics

Yes, they look relatively simple - but don't be deceived by their appearance.

For example, the choice of membrane material (its thickness, weight per unit area, elasticity, dimensional stability (Vs temp., etc.), electrical properties (dielectric constant, ohms/cm, etc.), the properties required of the step-up transformer, etc. etc.

Then comes the "stretching of the diaphragm over the rigid frame and permanently securing it in place, etc.

This is followed by the design and construction of the grid of charged wires. Thereafter, the design of the high-voltage power supply and the required wideband audio step-up transformer present their own challenge.

I could go on and on. How do I know - well, I was enamored with membrane loudspeakers back in the 1950,s and '60's, owned (and loved) a pair of Quad ESL-63's, and decided to design my own. Although I possessed a competent background in both physics and E.E., I soon discovered the project was beyond my capabilities. First, and perhaps most important, I lacked the necessary lab equipment and anechoic chamber for performing the necessary measurements.

Set back a bit by daunting aspects of the task confronting me, I decided to examine the properties of membrane loudspeakers in more detail before proceeding. After some considerable "text-book" research, I decided to abandon the project for the deeper I delved into the measurable and audible performance attributes of loudspeakers using a large membrane radiating surface, the more I came to realize that it was not as accurate a reproducer of complex musical transients, etc. as I had envisioned.

Don't get me wrong! It was fun learning how something that looks so simple and straight-forward can turn into a near nightmare of challenges. And, without wanting to purchase the necessary measurement equipment, e.g., precision microphones, chart recorder, a large number of expensive acoustic foam wedges for creating a small "anechoic environment (for measurements), etc. , I simply gave up.

Perhaps, there are those who have accepted the challenge and succeeded in designing and constructing their own ESL - but how did it sound and measure. Maybe good - but probably not as good as a used pair of ESL-63's! Or as good as a well-designed, conventional loudspeaker using a time/phase aligned array of drivers, fed by a first-order crossover, compensated for driver anomalies. (But here, too, an anechoic measuremnent capability is required to obtain documentably accurate performance.)

Anyway - look before you leap. ESL's look simple and their operation is relatively simple - but their design and construction can be a nightmare.

Anyone have a different experience? I hope so!

Best of luck,
John Dunlavy

Subject: Re: Most Accurate Loudspeaker?

Perhaps, we should give up our pursuit of "true accuracy", based upon a competent evaluation of a complete set of accurate measurements, combined with carefully controlled comparisons with live music, and attempt to locate an old non-electronic "Victrola" with a fresh supply of steel needles? (But where would we find a supply of old 78 RPM records?)

Hmmm!

Those who do not realize what a competent interpretation of a complete set of accurate measurements can reveal about a loudspeaker's ability to accurately reproduce complex musical waveforms, are simply uninformed about what today's true science and technology have to offer.

And, to give up chasing "perfect reproduction", although knowing it will never truly exist, seems a bit like saying we should no longer attempt to design and manufacture better cars, homes, TV sets, etc.

(But while engaging in the "chase", it is simply prudent to recognize that not all loudspeakers, etc. are created equal - with respect to true, documentable accuracy.)

Further, it seems to me that giving up the "chase" would make for an awfully boring and unrewarding life.

Best regards, John D.

Subject: Re: Wilson Watt/Puppy

To reproduce squarewaves (which consist of the fundamental plus odd-order harmonics (of gradually diminishing intensity) does not require that amps, loudspeakers, etc. exhibit bandwidths approaching infinity.

Indeed, to reproduce squarewaves (or other "complex waveforms) with an audible accuracy equal to that available from a system whose high-frequency response is flat to infinity, requires only that the audio system (amps, loud-speakers, etc.) have a bandwidth extending to the limit of human hearing,.e.g.,, 20 kHz., with a good modulus of amplitude vs. frequency and a good impulse response.

Since squarewaves, impulses, etc. are waveforms frequently approximated by the sounds produced by many musical instruments (and their combination within an orchestra), it makes good sense that all components of an audiophile system be able to accurately reproduce such waveforms.

Why not? Because limited audibility testing, conducted under questionable conditions, etc., failed to fully confirm that the accurate reproduction of complex waveforms was not essential?

Some good questionsare being asked here on RAHE - the answers to which I believe may affect the future vitality and integrity of High-End Audio.

There is an old saying: "Measurements don't lie - but measurers often do"!

Don't audiophiles understand that a lack of accurate measurements (especially for loudspeakers) is a "cover up" for hiding poor performance or the kind of performance that "can be heard but not measured"?

Why are most audiophiles so trusting when it comes to the accuracy of technical specs, performance claims, measurements, etc.? Are companies with little technical integrity "winning the battle" by convincing audiophiles that measurements are useless for determining the potential of a product to deliver accurate reproduction? Gulp!

And, where is our expensive government (Consumer Protection Agency, etc.) with respect to providing some assurance that highly advertised performance claims have at least a small measure of legitimacy? Obviously, none of us want our government to pursue a "witch hunt" for bad guys - but how about letting companies know they are "responsible for the accuracy of advertised performance claims, specifications, and measurements"?

And, one might ask, where are our cherished audiophile magazines when it comes to truly accurate coverage of equipment and loudspeaker performance? Don't they make enough money from advertising and the sale of mags to occassionally hire a technically competent engineer or physicist as a consultant to verify their opinions regarding product performance, etc.?

You would be surprised how many times I have had to answer the question, "If a full set loudspeaker performance measurements are important, why is your company apparently the only one to provide them and guarantee their accuracy?" Hmmm!

Perhaps, one answer might be for audiophiles to insist that manufacturers making claims for accuracy, etc. back up their claims with guaranteed accurate measurements, etc. - not merely subjective opinions?

And, yes! Some of today's best "digital room correction systems" can work wonders with really bad rooms - but they cannot fully correct for poor time-domain response and impulse smearing exhibited by a poorly designed loudspeaker.

Keep tootin' guyes and gals! There is still the old saying, "In the end, the truth will out!"

Subject: Re: Ported Spkr Bass Question

Just thought a few technical observations about "ported subs" might be interesting to you and other readers.

In physics, there is an old saw that says, simply, "there are no free lunches!".

If you begin with a woofer driver in a sealed enclosure whose volume more-or-less matches the requirements of the driver's electrical/mechanical properties, the bandwidth, the minus 3 dB frequency, the roll-off slope and the pass-band SPL can be calculated with reasonable accuracy using the well known methods published by Thiel and Small many years ago (not the Thiel who has a loudspeaker company). The low-end "roll-off" of a typical ported bass enclosure exceeds 24 dB per octave below system resonance.

Generally, the slope of the low-frequency roll-off of a sealed woofer enclosure, without internal absorbing material, occurs at a rate of about 12 dB per octave (after the 1st octave, or so).

If you add open-cell foam, etc. with good acoustical absorption properties, to the interior of the sealed enclosure, the "system Q" will drop, the overall efficiency will be lowered, the system resonance will drop and the overall system bandwidth will be increased (with fewer peaks and nulls in response). Again, these properties are readily calculable and measurable. These "improved properties" can be partly attributed to the simple fact that the velocity of an acoustical wave propagating through a lossy medium (such as open-cell foam) is lower than it is through air.

And, the presence of "lossy, open-cell material" greatly diminishes the amplitude of "standing-waves" within the enclosure (which often create peaks and valleys in amplitude vs frequency response).

If you take the same enclosure, sans the absorbing material, and create a "port", the efficiency will increase within that portion of the frequency range affected by the port. However, the laws of physics require that the "system bandwidth of the woofer-enclosure" be reduced. This results in the low frequency roll-off (below resonance) becoming much more rapid than 12 dB per octave, e.g., 24-30 dB/octave (or more) being typical.

This much more rapid roll-off actually reduces the amplitude level of radiation at very low audio bass frequencies - where bass energy is often "felt more than heard". Thus, while a port can add a "boost in efficiency" to a woofer-enclosure system, it will typically also reduce the amplitude of bass radiated at frequencies below resonance

So, the laws of physics always "exact a price" - nothing is ever a "freebie". Thus, you have a choice between a "slightly higher efficiency" with a ported enclosure (but with "a rapid roll-off in low-end bass") or a "slightly lower efficiency" with a sealed enclosure (but with an extended low-end response).

And, of course, the impulse response of a properly designed and damped sealed enclosure is virtually free of "over-hang and ringing", while the impulse response of a ported enclosure is hardly one to brag about.

Personally, being an "accuracy freak", I much prefer the accurate sound quality of a properly designed, sealed woofer enclosure.

Subject: Re: A question for Mr. John Dunlavy

Yes, as I thought I had mentioned in previous posts, we take a very large number of anechoic chamber measurements, both on and off axis.

As one would expect, the measured properties related to accurate reproduction do not significantly change out to about thirty degrees off-axis. Beyond that, the "path alignment" of the drivers becomes increasingly altered by the differential path differences. This results in an increasing degradation in the "off axis" properties of multi-driver loudspeakers.

Even membrane loudspeakers, having a single radiating surface, suffer off-axis performance problems because of directivity problems at frequencies where the width of the radiating surface becomes large with respect to a wavelength.

This is the reason that the manufactureres of most accurate loudspeakers recommend "toeing in" their loudspeakers such that they are pointed at and equidistant from the preferred listening location.

What we need is a small diameter dome radiator that exhibits flat amplitude/phase response over the entire audio spectrum and exhibits a high efficiency, good input impedance, etc., etc.

Hmmm!

John D.

Subject: Re: Another question for Mr. John Dunlavy

Locating a woofer driver on a large ground-plane, such that the outer edge of its cone is essentially flush with the surface of the ground-plane may not be an accurate means for determining the woofer's "anechoic response", etc.

When a radiating surface (woofer driver), with dimensions small with respect to a wavelength within the spectrum being measured, is located flush with a ground plane, it is radiating into "2 Pi space", which yields a "uni-directional" radiation pattern with a 3 dB "directivity gain" at frequencies where the driver's "radiation pattern" would be essentially "spherical" when mounted in the box with dimensions small relative to a wavelength..

One way to visualize this is to imagine a "spherical balloon" in free space with a radius equal to "1" (one). When the same balloon is mounted over a ground-plane such that the lower-half of the balloon is flattened, its new radius is increased by 1.414 times (as a hemisphere with the same volume). This difference in "radius" yields a "power gain" of two (2) or 3 dB.

This is elementary theory for antenna engineers who deal daily with such problems.

Burying a loudspeaker with its cabinet face flush with the ground will produce measurements that exclude the rather nasty radiation components attributable to "diffraction" from the edges of the enclosure.

Solving this problem led to my using acoustical absorbing material between affected drivers and the edges of the enclosure. Originally, AR claimed they had received the patent for this invention. NOT SO! I was the inventor and received the patent - not AR. (US Pat. 4,167,985) Hmmm!

Anyway, burying a loudspeaker with its face flush with the ground does not yield an effective nor accurate means for measuring the SPL, freq. resp., directivity patterns, etc. of a loudspeaker.

Nor will laying it with its back side flat on the ground solve the problem, because at mid and low frequencies, radiation from the drivers will refract and diffract from the enclosure edges, with considerable energy headed in the direction of the ground, where it will be reflected in forward directions, combine with the direct driver radiation and create peaks and nulls in freq. resp., exagerate impulse-response ringing, alter the shape of the step-response, etc. etc.

Best of listening,

John D.

Subject: Re: Dunlavy vs. The Tube

In comparing the measured square-wave reproduction of a loudspeaker to that of an amplifier, one must consider the different measurement protocols involved.

Measuring the square-wave reproduction of an audio amplifier is relatively simple, straightforward and unambiguous.

However, measuring the square-wave reproduction of a loudspeaker within an anechoic chamber is hardly a simple or straight-forward task. This is because most anechoic chambers, including the two large chambers at DAL, are imperfect with respect to internal reflections. In particular, they impose various limitations on the accuracy of frequency and time domain measurements at the normal on-axis listening distance of 10 feet, especially below about 200 Hertz.

Indeed, even relatively large chambers (like the two at DAL, 24 ft. long X 20 ft. wide X 16 ft. high) with the best, most efficient absorbing material (alternating, two-feet deep wedges of high-density, open-cell acoustical foam completely covering all internal services, exhibit some measurable reflections.

As these reflections become increasingly large in amplitude (relative to the direct-path signal), with decreasing frequency, they prevent taking highly-accurate "direct" measurements of some performance attributes, such as amplitude Vs freq. response, low-freq. square-wave reproduction, etc.

Indeed, measurements of a loudspeaker with "perfect" square-wave reproduction would reveal some "ripples" and a "tilt" at the top and bottom of square-waves at frequencies below a few hundred Hertz, due to these reflections from internal chamber surfaces.

Thus, when viewing "anechoic chamber measurements" of loudspeaker square-wave reproduction with those of audiophile amplifiers, the "chamber anomalies" must be taken into consideration.

When this consideration is factored into the "comparison", the reproduction of DAL loudspeakers is found to be comparable to that of any tube-amps with the same frequency response attributes.

Thus, in this context, the square-wave measurements of DAL's physically small SM-I loudspeaker, sent to a few requesting them, are not too "shabby".

Best of listening,

John D.