JANSZEN

Model One Electrostatic Hybrid Loudspeaker

White Paper

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Introduction

So, we have a fine old name, our engineering is done primarily by a seasoned engineer who is a son of the inventor of the first commercially practical electrostatic loudspeaker, first used in the JansZen 1-30, which helped get the whole hi-fi thing rolling in the first place more than 50 years ago, not to mention that the older products that were based on that original invention were among the most revered and reliable ever.

But it has been over 30 years since the last new thing from JansZen, and speakers in general are much better than they used to be. You might be wondering what, after all these years, could JansZen be doing that isn't already being done better by someone else?

Of course, we think that's ridiculous, because we think we're the best, but so do all the other audio companies, so if you have your doubts, please read on.

JansZen Model One

Three-way System

In the JansZen One, a dynamic woofer handles the subsonic and lower bass range from 20 Hz to 200 Hz, an electrostatic line array handles the upper bass and lower midrange from 200 Hz to 1 kHz, and a second electrostatic line array handles upper mid-range and treble from 1 kHz to 20 kHz. Our three-way designs overcome the problems of wide-range or full-range, single membrane, electrostatic systems without introducing drawbacks, and our selection of sound field shapes and crossover points helps produce quintessentially natural sound throughout a wide listening area.

It may seem surprisingly simple, but the tweeter achieves wide dispersion (80° -3 dB at 10 kHz) just by being physically narrow. The principle is that smaller apertures have wider dispersion at a given frequency. For a line array, only the width has to be made small, which creates a vertically confined beam with wide horizontal dispersion and more than adequate SPL's.

This configuration allows the tweeter to be open, single-faced and physically flat. As a result, it produces a more uniform sound field than other means of creating wide high frequency dispersion, such as facets, lenses, horns, and membrane curvature. It also improves gap uniformity relative to curved construction for reduced harmonic distortion.

We placed our tweeter crossover point at 1 kHz to keep it below the frequencies critical to locating sound in space, and we use a very soft, zero-compensated crossover filter. With most speaker systems, the tweeter crossover point is usually placed in the most sensitive part of this band, where the associated phase skew smears the sound stage. It can also be partially responsible for an unnatural aspect to the sound, sometimes described as a "veil" separating the listener from the music.

Locating the midrange panels immediately adjacent to the tweeters (in fact in the same physical panels) prevents dislocation of coincident mid-range and high frequency sound. Locating the woofer near the floor and cutting it off at 200 Hz takes what would be a problem with floor interaction, and turns it to advantage. It does this by using the floor to blend the woofer's floor-modified spherical wave front seamlessly with the electrostatics' height-limited, cylindrical front, at least within the confines of an indoor listening area. Lastly, only low frequency sound that can not be vertically located by the ear is emitted by the low-mounted woofer, so there is no tendency for the apparent height of the sound to change with frequency.

Using separate electrostatic drivers makes the amplifier load well behaved, because the high capacitance of a large transducer is not driven at high frequencies. (*Among past makers of all-frequencies-to-all-membranes systems, Acoustat got around this problem by clever use of two transformers with a crossover filter connected between their primaries. Interestingly, a claimed advantage was the lack of a crossover.*)

One of the most noticeable things about the appearance of these speakers (shown below without the grille in place) is that the enclosure that houses the electrostatic panels is tilted sideways relative to the primary enclosure. This provides a continuously varying edge distance, and is done to prevent edge diffraction interference from occurring at any particular frequency, rendering it inaudible.

Tilting works only with a line array, and accomplishes the same thing as the more elaborate diffraction reducing features that are sometimes used with point source drivers (spherical enclosure, backward flares). The tilt is in the same direction on both sides so the wave-front axes cross in parallel. (*Line arrays produce cylindrical waves, which have an axis of symmetry*.)

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Closed Back, Monopolar ESL's

Most people expect a dipole when they hear about a new ESL, and are surprised by the design of the JansZen One. Under the right conditions, dipoles probably have an advantage over fully enclosed speakers in their potential, rarely achieved, to produce the most diffuse, uniform sound field, and produce more overall sound above their lower cutoff frequency. For these reasons, we will be offering dipoles at some point.

On the other hand, dipoles, especially large, wide-range dipoles, are not for people who are interested in a speaker that they can just plant in their living room and get good results. They are better for when one's primary hobby is refining the stereo system. We love these guys, but our first product was designed for the other kind of audiophile.

JansZen One, Back View

Convenience was a major consideration, and is also a reason why amplifiers are built in. Closed back line arrays can be placed half a meter (twenty inches) from the back wall, and diffusion of the sound is similar to that from point source dipoles, and far better than from point source monopoles. This is because of the radiation characteristics of the line array, where much higher sound levels reach the walls for redistribution than from point sources. You can have the installation tech put them where you want them, check their response and adjust their tweaks if necessary to match their positions, connect an iPod console or a decent CD player with a volume control, and that's that. You get home from work, hit play, and escape into musical paradise.

Dipoles can represent an inconvenience within a living area, can be a decorating challenge, and can be difficult to set up for optimal results. To retain all the benefits of dipole radiation, they generally require at least six feet (about 2m) to the rear wall, which gives them a large footprint. Their rear radiated sound field interacts strongly with objects in the room, and this can necessitate changes in the positions of other objects and possibly the addition of potentially visually unharmonious acoustical treatments, namely absorbers, diffusers, and maybe even traps.

The other drawback of dipoles is front/rear cancellation of low frequency sound, which can be thought of as the less directional, longer waves going out the front and getting sucked in the back, and vice versa. This moves the cutoff frequency up for a given radiator area. With ESL's of any practical size, this forces use of a rather high cutoff frequency on a dynamic woofer to fill in the missing bass and sub-sonics.

Going too far up in crossover frequency will cause frequency dependent movement of the sound source location, which decreases naturalness. Also, dynamic speakers have more distortion than ESL's, and are practically impossible to make behave well at frequencies above a few hundred Hertz (cone breakup, secondary suspension resonances, etc.). Even though the distortion levels can be kept well below what is audible, less distortion causes less fatigue and sounds more natural. Lastly, at higher crossovers to the woofer/subwoofer, blending of the sound fields from the ESL's and the woofers becomes problematic and decreases sound field uniformity.

In the One, the crossover is at 200 Hz, which is low enough to prevent vertical dislocation, and since each speaker has a woofer, there is no horizontal dislocation problem in the woofers' upper octave. Also, the floor is helpful in blending the sound fields, rather than causing detrimental diffraction effects.

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ESL Panels

Our ESL (electrostatic loudspeaker) panels were newly developed by David A. Janszen. The basic principles are familiar, but marked improvements were made in materials and construction.

The stator electrodes have no sharp points exposed that would cause ozone-producing coronal discharge.

The stator frames have a coefficient of thermal expansion that matches that of the electrodes. This prevents mechanical deformation when undergoing the temperature extremes that are normal during shipment. Such deformation would otherwise alter the gap widths, causing reliability problems and increased harmonic distortion.

The JansZen ESL Panel, showing low Tc injection molded stator fames and electrodes. The tweeter and two midrange elements are integrated into each unit.

Bias and drive voltages are about half of what they once were for equivalent loudness, which further enhances reliability.

The adhesives are modern urethanes and epoxies. Wiring is finely stranded, silver plated, and Teflon insulated. Membrane connections are made with silver-filled epoxy. Membrane charge distribution is done using a modern, high resistivity coating.

The ESL's are mounted in their own, separate, sealed sub-enclosure within the primary enclosure, rather than being operated as dipoles. They are made this way mainly to reduce interaction with the room, which makes them easier to set up, and to allow placement nearer to back walls, so they take up less space. It also circumvents front-to-rear cancellation at lower frequencies, which importantly allows the midrange panels to be operated in the upper bass region, which in turn allows the woofer to be rolled off from a quite low knee of 200 Hz. Aside from the fidelity benefits of having ESL's covering the upper bass region, this is low enough to get around woofer response uniformity problems that would otherwise be caused by floor effects.

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ESL Membranes

The membrane and membrane coating materials are immune to the effects of ozone and humidity, so there is no reason to be concerned about long term reliability in extreme climates. They also provide an unusually good impedance match to the air load.

The membranes are constructed so as to hold off arcs. Of course, there is a limit to how much voltage can be held off without making the membrane too heavy, but it is more than enough to prevent arcing in the active speaker systems, where amplifier power is not high enough to break down the coating.

The situation is different in passive versions of the systems. The sound of ESL's does not exhibit distortion as their limits are being reached, so there is no audible warning that a person can listen for. In contrast, EDL's (electrodynamic speakers) produce distortion that sounds "too loud" when they are heating up.

For protection of our speakers, we've included circuitry that monitors the drive voltage without creating any interference. A display light indicates when the voltage limit is being approached, going from green to yellow to red. If the voltage limit is exceeded, the monitor automatically pads the sound down by about 20 dB for about ten seconds while causing the display light to flash red. Our amplifier power recommendation is a maximum of 50W RMS, but one may use the overload indicator light to stay within the safe range.

Padding is done actively by seamlessly connecting a load steering network that sidetracks some power away from the ESL's. It actually improves the amplifier load when activated and completely avoids stressing the amplifier during an overload event. The circuit resets itself, so there is no need to do anything but turn down the volume. Our padding method is substantially better than any other, such as a fuse that would stress the amplifier by creating an open circuit and degrade the sound by creating some power compression near the limits, or a strapping circuit that would stress the amplifier by increasing its load. This circuitry is also one reason why the passive version costs the same as the active one.

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Woofer

An ultra-low distortion, long-throw electrodynamic (ED) woofer is used for the lower part of the bass range, where the phase effects associated with this type of driver are inaudible. In the JansZen One, the woofer covers the range from about 20 Hz to 200 Hz ($\pm$3 dB). It has a single piece, unvented, lightweight, ultra-stiff aluminum cone that exhibits no breakup when reproducing these low frequencies. Its surround is butyl rubber, which will never require replacement.

Its maximum throw is nearly two inches, but it is driven only in its more linear short throw range. A number of proprietary features reduce its power compression, inductance, and eddy current generation, thereby providing exceptionally good motor linearity, which in turn produces inaudible levels of distortion throughout its operating range.

A low crossover from midrange to woofer is possible in a JansZen hybrid because our unified upper bass / midrange ESL panels exhibit an extended low frequency range for their size. The low crossover frequency helps make constructive use of the floor of the listening area, which serves to shape the wave front from the woofer to blend well with the wave front from the ESL's. A higher frequency could cause problematic effects from the floor.

The woofer enclosure is sealed, which improves linearity and thereby lowers distortion. The woofer amplifier, which has an extremely low output impedance, is relied on to supply most of the damping. This combination provides transient speed without resonance or overshoot, which is the key to "tight" bass. Although the size of the woofer is irrelevant to this "speed" characteristic, a large woofer area is required for supplying deep bass extension without distortion. A number of smaller woofers might have been used instead, but this would have added to the total moving mass and thereby reduced the "speed", and the shorter throws would have increased distortion for a given loudness at lower frequencies.

Due to the extended low frequency response of the ESL's and the exceptional speed and low distortion of the woofers, our hybrid systems do not represent a sonic compromise relative to full-range ESL's. At the same time, they offer a considerable size advantage, as well as bass extension and subsonic impulse delivery that simply can not be provided by ESL's of any practical size. In addition, the very high voltages needed for electrostatic woofers (or full range, single membrane systems) are not present.

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Crossovers

There is an analog front end that separates the signal into three channels, one for each amplifier. Among other things, it performs the basic crossover function by limiting each signal to a band that is appropriate to the associated driver. Relegating this function to line level circuitry lets us accomplish several unusual things aside from the obvious elimination of the large capacitors and inductors found in passive crossovers.

The tweeter crossover point and design minimize phase skew in the band that the ear uses for discriminating position information, which is critical to providing the highest level of detail in the stereo image as well as the most convincing nature of the overall sound.

The transitions between drivers are carefully tailored so that there is minimal, inter-driver phase interference, which avoids inherent response nodes that would reduce sound field uniformity.

The listener is supplied with some control over the response. Bass and subsonic response controls are provided for room and preference matching. There is also a tweeter roll-off control that places an adjustable upper knee on the response as a way to "dial in" the response to match the treble attenuation of a concert hall at various amounts of distance from the stage.

Lastly, each speaker system has a 180° phase switch for those who wish to alter the absolute polarity. Changing the absolute polarity means setting both speakers to 0° or 180°, not in opposite phase to one another. Of course, the effect of absolute polarity has been the subject of controversy, but we feel that there are technical reasons why there may be something to it. (No one thinks the phases should be set opposite to one another, which would cause serious problems with frequency response, sound field uniformity, and sound stage presentation.)

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Amplifier Primer

In the JansZen One, there is a separate amplifier for the ES tweeter, the ES mid- range, and the ED woofer. All three are variations on a topology known generally as class D. Class D for audio has come a long way since the class D concept was invented by Peter Baxandall in 1959. It was chosen here for its capabilities of high fidelity and low heat generation, allowing all the electronics to be sealed into the speaker enclosure while still producing a high fidelity, high power output.

The class D designation covers a wide range of implementations, and as may be obvious to many, there is also a wide range of acceptance and understanding, even among practicing electrical engineers. This is mainly because, given the range of implementation possibilities, it is impossible to generalize widely about the topology, and difficult to have two people talking about the same thing when they try. Adding to the confusion is a recent proliferation of complete, class D, amp-on-a-chip IC's for low to medium grade audio applications, some of which can put out a surprising amount of power.

For novices, it is important to mention first that the class of an amplifier is not related to its grade or quality. Also, the 'D' in "class D" does not stand for "digital". The class indicates something technical about how an amplifier works, and with power amplifiers, it indicates in particular how the output stage works.

With classes A, AB, and B, the output stage directly produces power that is proportional to the input voltage with no further processing. When the input voltage is halfway to its full scale, the output power is halfway to its full scale. This continuous variability is called "linear" operation or "linear mode", which can be confusing, because it is never perfectly linear in the strictest sense. Class G is also a linear class, but with stepped or variable supply voltage to improve efficiency by reducing dissipation in the output transistors. There are other classes not in commercial use for audio.

Linear operation has three main drawbacks: Firstly, it causes a lot of the power to be converted to heat, because the output devices have to act somewhat like varying resistors to provide a continuously varying output. This is not a problem for most audiophiles, but surely keeps these amplifiers out of our sealed enclosures.

Also, when the input goes, say, from quarter-way to halfway, the output may go from something slightly different than quarter-way to something slightly different than halfway. This is known as non-linearity and is the main cause of harmonic and intermodulation distortion in such topologies. Lastly, with a push-pull output stage, as the continuously alternating signal crosses zero from positive to negative and vice versa, and the power is passed from one output device to the other, there is a slight glitch known as crossover distortion. Any difference in the gains of the two devices also produces waveform asymmetry that causes distortion. All these contributions to distortion can be minimized but not eliminated by using feedback, hand-picked device matching, and bias control.

The basic class D topology uses a pulse width modulated (PWM), switching output stage. The output transistors are either on or off, never driven in their linear ranges. This means that heat generation is minimal, but more importantly to audiophiles, there are no in-band crossover artifacts, and no audio distortion is caused by output transistor non-linearity.

The lack of crossover artifacts is analogous to the operation of single ended linear mode amplifiers, but in any type of linear mode amplifier, there is no way to head off the effects of output device non-linearity, so it can only be suppressed in such amplifiers by using feedback after the fact and minimized using careful part selection and biasing. With class D, it is never produced in the first place.

In class D amplifiers, the audio signal is recovered from the PWM signal using a low pass filter with its knee well above the audio band and well below the switching frequency, meaning usually an octave or so above 20 kHz. In so called "filterless" class D designs, the inductance of speaker itself is used as the filter, but this is unsuitable for high fidelity applications.

Some class D amplifiers take a digital input and use it to vary the pulse width, some digitize an analog input and use the data to generate the PWM, and some convert the analog signal directly to PWM. Our current systems are of the last type, which is non-digital, maintaining the signal in analog form until conversion to PWM. The first two types *are* known as digital amplifiers. None of these three approaches is necessarily better than another, and the results depend on the specifics and quality of the designs.

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Our Amplifiers

There are several high grade, class D amplifier implementation types. We use one of these for the two ESL amplifiers, and another for the woofer amplifier.

The ESL amplifiers use an analog-input controller that sends precise, complementary waveforms to an open loop, full bridge output stage based on extremely fast, low Rds, power FET's. The fidelity of a less well implemented open loop class D amplifier would suffer from offset, gain, and phase errors in the conversion circuitry. In that case, complex feedback circuitry would be needed to reduce the consequent distortion, but our use of exceptionally fast FET's mated to the optimized, integrated controller makes it possible to achieve low distortion from our amplifiers without feedback. The one remaining drawback to open-loop operation is that there is practically no power supply rejection, so a well regulated and filtered supply is used for these amplifiers.

LC (inductor-capacitor) output filters are employed to recover the audio band signal. The ESR (equivalent series resistance) of the inductors is not a problem in our case. That is because with ESL's, there is no mass to control, which makes damping factor irrelevant to attaining good transient response. The inductors were specially developed to avoid introducing distortion from hysteresis and saturation effects, yet being toroidal, they still contain their magnetic fields and prevent EMI.

These amplifiers have about a 115 dB noise floor and less than 0.1% THD+N from 1 W up to about 40 W (80% of full power), climbing to 1.0% only at full power. At low power, THD increases somewhat, but is still less than 0.4% at 100 mW. This amount of distortion at low power is inconsequential because the distortion products are reproduced at such low absolute sound levels that they are inaudible.

Once distortion issues are out of the way, the most important factor for natural presentation of subtle or high crest factor material is the dynamic range, which is more than 100 dB from these amplifiers. This wide range works in concert with the high linearity of the speaker elements, allowing extremely quiet sounds that only require power as low as 400 μW to be reproduced in uniform proportion to louder sounds that require up to 40 W, considerably surpassing the dynamic demands of even the best CD material.

The woofer is motivated by an unmodified, 500 W ICEpower module from Bang & Olufsen. This amplifier is powered by a linear supply that uses a large toroidal power transformer to eliminate stray fields and mechanical hum. ICEpower amplifiers are all exceptionally well designed and widely accepted, long proven in a variety of applications. B&O uses innovative, patented feedback methods to provide low distortion, good power supply rejection, and extremely low output impedance. (*Feedback is required in high power class D amplifiers because there are no suitable, high voltage, power FET's yet developed that are fast enough to run open loop without excessive distortion.*)

Because of the low output impedance, the damping factor in combination with our woofer is a whopping 800, providing exceptional cone control and contributing to the high "speed" (well behaved transient response) and extremely low distortion of the woofer portion of the system.

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Data

It is never a bad idea to have a look at some curves, even when generated by the manufacturer. Ours were run using the following equipment: Sampling and signal generation were supplied by an external "sound card" running at 96 kHz / 24-bits and generating pink noise. The card was connected directly to the speakers with single ended cables. A high resolution, 24-band per octave, PC-based spectrum analyzer made the measurements, after which the curves were smoothed to reduce jaggedness caused by the room. The input transducer was a high grade, omnidirectional, free field, condenser microphone whose innate response curve was compensated up to 20 kHz by the analyzer. A garden variety phantom supply provided bias voltage to the microphone. The microphone cutoff was 20 kHz, so the measured amplitude above this frequency fell off at least partially for this reason.

The testing area was an actual living room that had no special acoustical properties or modifications, such as diffusers or absorbers, and was in fact rather challenging acoustically, with many large cavities and reflective surfaces. The effects of fundamental room modes and the nearness of the back wall (sheet rock) to the microphone are evident in the plots.

The room's overall dimensions were about 24' x 15' (7m x 5m) with a fairly high, sloped ceiling, two open doorways, two open staircases, two couches located a couple of feet away from each of two walls, a carpeted floor, lots of glass windows with no drapes or shades, one brick wall, and a long, solid concrete bench cantilevered off the brick wall.

The speakers were about 7' (2m) apart, center to center, facing across the short dimension of the room and located well off the center of the long dimension, under the lower part of the ceiling, and flanking a narrow, open, fireplace set into the brick wall.

The backs of the speakers were about 19 inches (48 cm) from the walls, which was as close as possible due to the presence of the concrete bench running along that wall. A closer position would have affected the frequencies of the humps in the bass response, but not the midrange or treble, as it would have with a dipole radiator.

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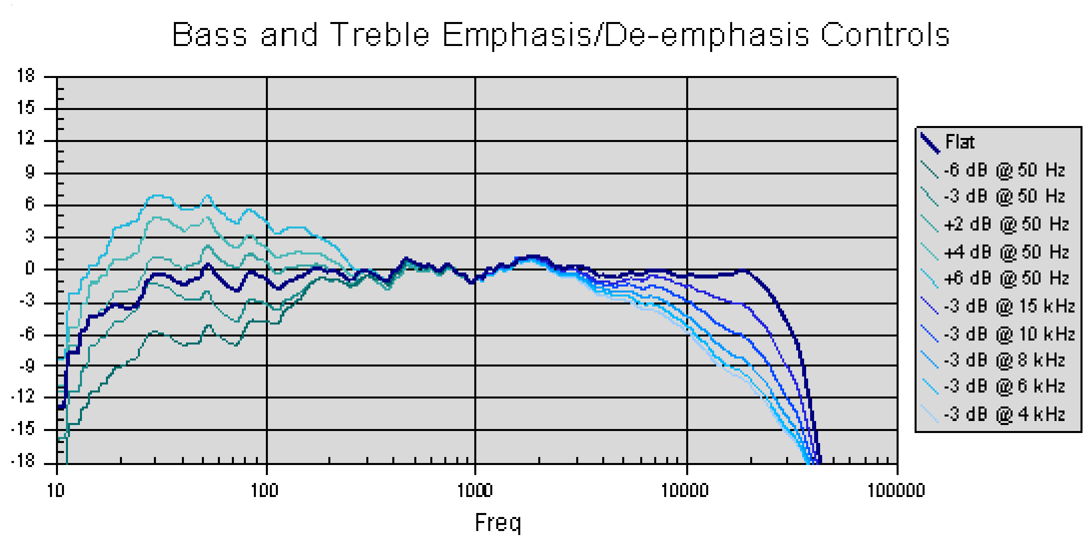
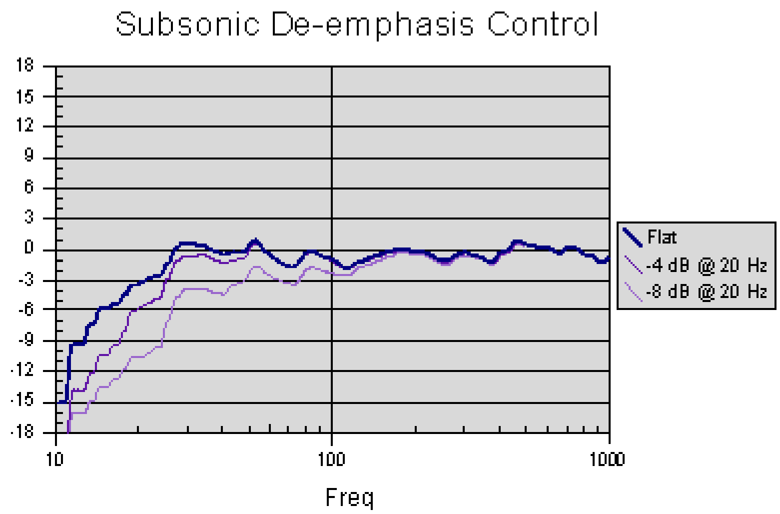
Effects of the Response Controls

The first set of curves shows the frequency response at several settings of the controls. Measurements were taken from one speaker, on axis, from 10' (3m) directly ahead of the speaker, with the microphone positioned about where one's head would be over one of the couches. The rear wall is about 1m behind the microphone. Room effects and numerical instability in the DFT (digital Fourier transform) processing made the bass plots somewhat inconsistent, but the basic functions are apparent. Note the overall frequency response, with usable output from 10 Hz to 30 kHz.





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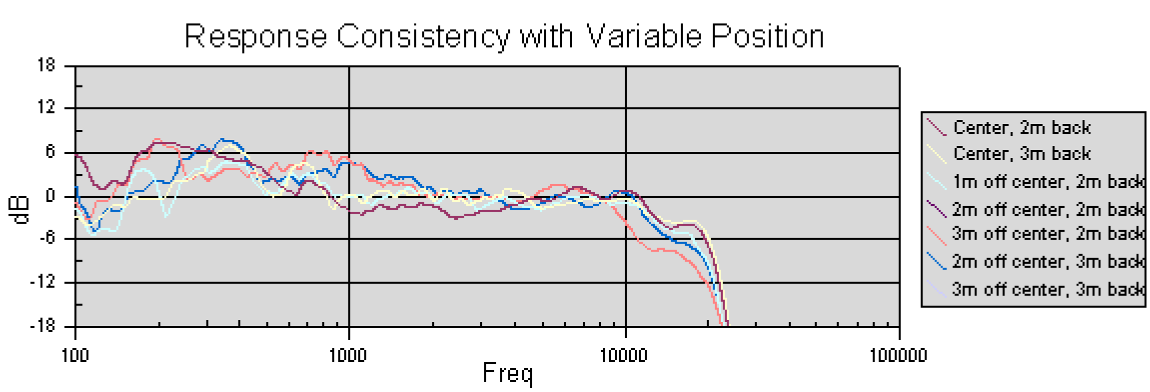
Sound Field Uniformity

The following set of curves helps demonstrate how consistent the sound is when the listener moves around the room. This consistency conserves the stereo image and the quality of the sound reproduction.

Note that at 1m off center, the microphone was directly ahead of the right speaker, and at 2m and 3m off center, the microphone was 1m and 2m *outside* the area between the speakers. The microphone was never placed directly on axis of either speaker, which is why the very top octave is down a bit even in the central positions. The effects of the inevitable standing waves are noticeable but not predominant in the plots, and in any event are not audible when listening.

An important effect of the sound field produced by line arrays is that as one moves away from the array, the sound that arrives directly from the speaker drops off half as quickly as it would from a point source.

It is this low rate of drop-off with distance, along with the wide treble dispersion, that are the primary factors in maintaining the stereo effect and the response flatness. Of course, the room's impulse decay time and the presence of a large number of reverberant modes also help maintain response flatness by producing adequate diffusion. The same property of low drop-off rate, though, increases diffusion for a given room impulse response time. This has the benefit of allowing a room with less reverberancy and hence less image blurring to provide adequate diffusion.



The measurements were taken with both speakers running. The speakers were 2m apart, center to center, and canted inward by about 15°, aimed at a point a few meters behind the usual 3m back listening position. The microphone height was about 1m. The more distant, off center curves were offset by either 3 or 5 dB to demonstrate conservation of response flatness and improve the legibility of the plots.

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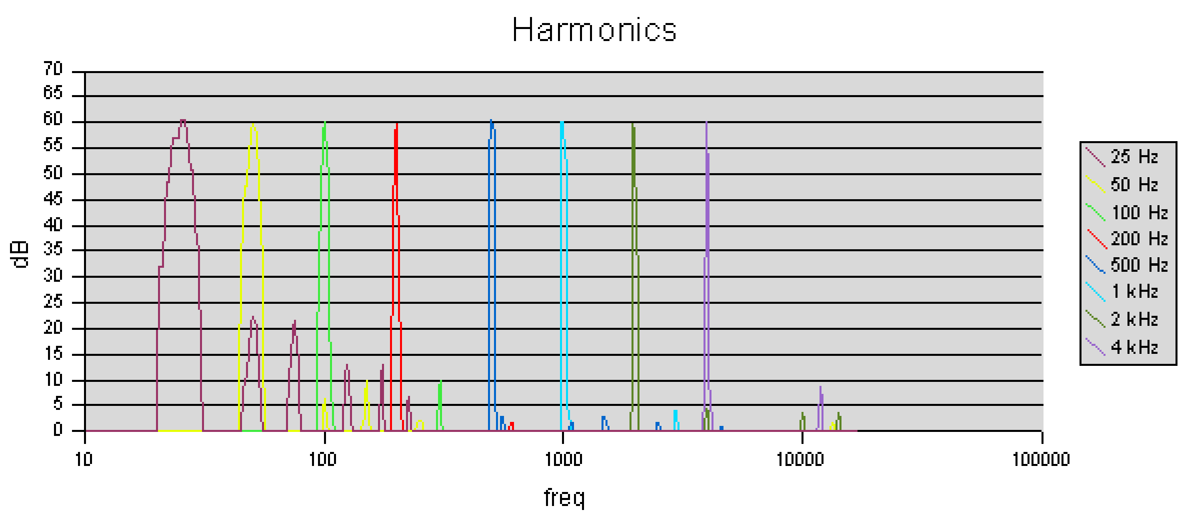
Low Harmonic Distortion

The following plots show the harmonics generated with a sine wave input at several frequencies reproduced by each of the three speaker elements (woofer, midrange, and tweeter). Note that these plots represent the combined distortion contribution of the amplifiers and speakers. The low distortion levels demonstrated by the ESL panels in the mid and treble ranges are very difficult to achieve with electrodynamic speakers.



We hope that this data does not take attention away from the primary benefit of electrostatic speakers, which is the natural reproduction allowed by their fast transient response and lack of coloration and breakup. We present it because low distortion is important as well, particularly since distortion causes listener fatigue, even when some- what below audible levels. This is why fatigue is often lamented by owners of purely electrodynamic speakers, even when their systems sound great. JansZen speakers do not cause listener fatigue.

Few manufacturers are willing to publish their distortion figures, but among those who are, such as Aura Sound, Inc., you will not find such low numbers. THD from our systems is 0.16% at the commonly quoted frequency of 1 kHz, and less than 0.40% anywhere in the entire spectrum above the very lowest frequencies, below which the woofer cone really gets moving and its suspension non-linearities come into play.

At 25 Hz, it is obvious that there is a substantial increase in THD as nonlinearity in the motor and surround begins to play a part, but it is inaudible and non-fatiguing for three reasons: 25 Hz is below the frequency that humans can hear as "sound", which is why it is called sub-sonic; THD is still less than 2%, which is low for any woofer operating in the sub-sonic range; and the harmonics have a nice, natural profile that mimics those from natural sources, such as elephants and huge organ pipes.

We made no attempt to find and present frequencies with the best numbers, the point being that this would not have been beneficial anyway, because the performance is fairly constant.

It is often said that THD below 3% is inaudible at any frequency, but this actually depends on the amplitude profile of the harmonics, and on the fundamental frequency. In general, a profile with an abundance of odd harmonics is more easily heard and sounds less pleasing than when the harmonic content falls off gradually with order, including even harmonics in "natural" proportion, as does the sound from musical instruments. It may be an academic point, but even if the THD at 1 kHz were 3% from JansZen speakers, namely about twenty times higher than it is, the proportion of even harmonics would be high enough to avoid a displeasing result, although fatigue would then become an issue. (*RTR once made tweeters based on the JansZen design with drastically altered design parameters that were about 10 dB louder than the original design, at the expense of having 3% THD. The initial impression was pretty wild, but then their harshness became grating.*)

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Horizontal Dispersion

The following table of outdoor horizontal dispersion data indicates about an 80 degree included angle at 10 kHz, and about 25 degrees at 20 kHz. The original spec of -3 dB at 66 degrees at 10 kHz was a conservative theoretical value, based on the tweeter geometry.

	Angle				
	On axis	**10°**	**20°**	**30°**	**40°**
10 kHz	0.0	-1.3	0.0	-0.3	-3.4
20 kHz	-0.8	-2.3	-8.2	-13.8	-18.8

Response [dB] at various angles relative to axis

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Transient Response and Phase Coherency

A fairly decent indicator of these properties is the ability of a speaker to reproduce a square wave. This reveals the rise time, the presence of phase skew, and the presence of ringing or resonance. We chose 1 kHz, where all the harmonic content that must be reproduced in proper phase and amplitude is within the range that the ear uses for determining position. The measuring apparatus is sensitive to harmonics up to 10th order at full amplitude. The horizontal scale is 500 microseconds per division.

The plot below demonstrates an aggregate rise time of less than 100 microseconds, which is the limit of the instrumentation. It also shows excellent superposition of harmonics, and a complete lack of underdamped behavior.

The ripple is mainly the result of cavity effects between the microphone and the speaker, which selectively suppress or reinforce some of the harmonics.

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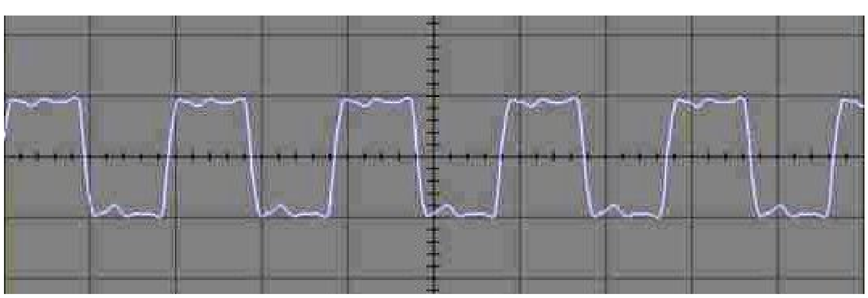

Summary

The aggregate system is free of audible distortion and is exceptionally linear throughout its very wide amplitude range. Details and ambient subtleties are thus fully revealed no matter how well hidden beneath the more prominent program material, and fatigue never sets in.

Phase is kept true in the ear's critical position sensing band, so the sound stage is detailed and the image contains the original instrument positions as recorded.

Sound pressure levels are high enough to reproduce performance dynamics faithfully.

The stereo image produced by the cylindrical sound fields is wider and more uniform than that of the spherical fields produced by point sources. This reduces the effect of listener position on the stereo image, providing a convincing sound stage throughout the room, even in positions that are not between the speakers.

Because of their fully enclosed construction, they are more space efficient and easier to set up than dipoles.

The frequency response can be adjusted to suit your preferences and match your listening room.

This adds up to convincing reproduction of recorded sound throughout your room, and that translates into consistently strong musical pleasure for you and those with whom you share the experience.

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Invitation

The sound of these systems speaks for itself better than any compilation of technical details. Please contact us to make arrangements for a private audition. An enjoyable musical experience like no other awaits.

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JansZen Loudspeaker, Ltd.
614-448-1811
www.JansZenAudio.com