Audioholics 2010 Subwoofer Shootout Measurements Overview

by paul — December 23, 2010
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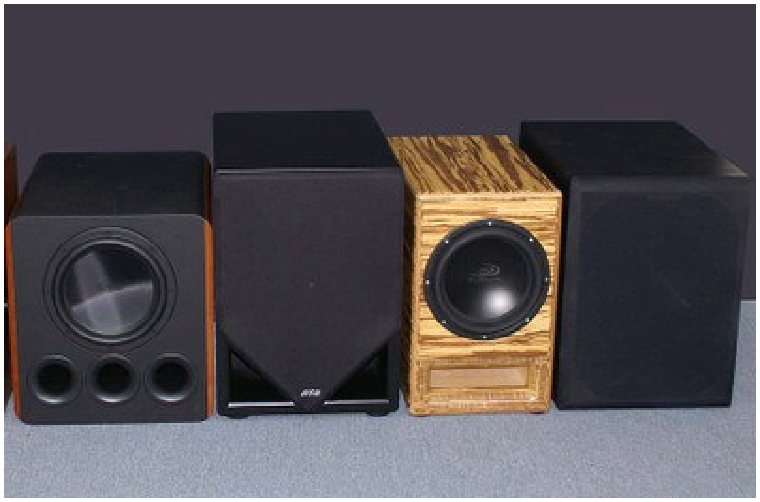

Subwoofers in our shootout

The goal of our most recent subwoofer shootout is to give us as consumers, a reliable and quantifiable way to purchase home theater subwoofers with some confidence that the products we hear and read about are in fact well designed and capable of sufficient output at low frequencies to reproduce both music and movies with the impact and realism that you expect from a well designed and executed product. The products in this comparison were chosen based on a box size of 6 cubic feet or less, and a maximum retail cost of $2000. This article discusses our testing methodology for measuring all of the subwoofers.

Product Entries

Brand	Model	Box Dimensions	Dimensional Volume	Retail Price	Product Description
Rythmik	FV15HP	18" x 24" x 24"	6 ft^3	$1,300	15" ported servo with 600 watt amp
Funkywaves	FW12.X	24" x 24" x 14.5"	4.83 ft^3	$1,950	12" ported with external 2000 watt amp
SV Sound	PB12-Plus DSP	25" x19" x 21"	5.77 ft^3	$1,399	12" ported with 800 watt amp
HSU	VTF-15H	25" x 18" x 26"	6.77ft^3	$879	15" ported with 350 watt amp

Putting your products in the hands of a stranger whom you do not know is risky business. There are more than a few folks in audio who are more hobbyist than engineer or scientist. Having your flaws displayed publicly requires a lot of courage. I can say with a clear conscience, there were no perfect entries. For a certainty there were excellent, fair and mediocre products. Some were so mediocre in fact, they withdrew from the competition. When viewed in that light, keep in mind those left here, are here because they were the best of the batch. The few people in the audio business who do know me well will not tell you I am critical. They will tell you I am VERY critical. My job is basically finding flaws, eliminating them and designing a better product. A product which may make the typical consumer ecstatic, may draw faint praise from me.

Some years ago when subwoofers were becoming much more popular, while walking the show floor at CES, (the Consumer Electronics Show) I acquired a loudspeaker component (woofer) catalog from a nondescript Chinese driver vendor, who had gone to the trouble of identifying every single one of the woofers pictured in his catalog as “subwoofer”. In fact, among the dozens of woofers pictured, regardless of the application, or range of usable frequencies covered, they were ALL identified as “subwoofers”.
The use of the term is not necessarily a guarantee you will be buying the real thing. Please note; there were NO fakes in this reviewer's opinion in our shootout. All the products submitted represented the best ability to produce, design and source parts, of all the contestants in the shootout. That does not mean there were any perfect entries, or flawless products. Pretty much every decision you make as a loudspeaker designer requires a compromise. The reality is, some folks are better at making choices and compromises than others, and I am hopeful that the consensus of opinion of our readers will be that this reviewer has done his best to be fair and even handed with all entrants. There were things I liked and disliked about all the subwoofers. Those shall all be disclosed in the individual reviews that follow.

Is This Test Fair?

We can expect the participants who yield the best results (loudest outputs within the distortion ranges allowed) to say yes, very much so. Those entrants at the bottom of the maximum undistorted output table may say no. My opinion is not really, and here is why. The boxes are of different sizes and different prices. The companies will follow their own ideas and formula's for product creation, and some were clearly better equipped from the standpoint of resources and know-how than others. On the flip side, it's very difficult to find competitive products of exact box size and retail price, but we did our best to collect similar priced samples from willing participants. Still, the main limitation beyond amplifier power for system output is actually box size and tuning frequency (the lower the tuning frequency, the less efficient the subwoofer is, holding box size a constant). Since this is NOT controlled, the larger boxes have a considerable advantage over the smaller sized boxes. The advantage of the smaller size box is clearly in room placement location and wife-acceptance-factor. (Universally known and hereinafter referred to as WAF among speaker builders). Keep in mind Internet buyers tend to be less concerned about box size and more concerned about product performance, hence why many of the Internet direct brands have the larger box sizes compared to their brick and mortar competition.

Pictured (left to right): SVS PB12 Plus, HSU VTF-15H, Funkywaves 12.x, Rythmik 15VHP

CEA 2010 Subwoofer Testing - What Is This and Why We Use It?

For many years, a traditional means of quantifiably measuring distortion has involved the very simple principle that if you excite a device under test with a single frequency, any other frequencies which it simultaneously produces must therefore be distortion. As a result of measuring this repeatedly, on all different kinds of gear we find that our reproduction devices tend to add multiples of the excitation frequency, called simply enough, harmonic distortion. (If you excite a speaker with a 1000 Hz signal, and it simultaneously produces 2000 and 3000 Hz, it has created a 2^{nd} and 3^{rd} harmonic of the fundamental frequency, or simple harmonic distortion. It has been know for quite some time by those attempting to relate the simply derived number which measures only the amplitude of those harmonics relative to the fundamental, known as THD or total harmonic distortion, correlates quite poorly with the human perception of how badly the original signal is distorted. The 2010 CEA standard addresses this issue by using a progressively more stringent limitation on the allowable distortion. Higher-order and/or odd-order harmonics have progressively lower allowable distortion limits, as these tend to subjectively be more offensive to the human ear than lower order and/or even-order harmonics. I have personally seen, as far back as the early 1990's from my own research as a subwoofer system designer for Miller & Kreisel Sound, that we often could "tweak" a compressor in such a way where we found the result more musical and pleasing, yet the THD as measured by some very expensive and reliable HP equipment would actually go up as a percentage of the output. In my own personal experience, this was almost always a trade-off for more low harmonics (notably second or third) for less higher harmonics (fourth or higher). We shall discuss this relatively new CEA 2010 standard in some detail, and hold true to it where feasible.

The Purpose of This Subwoofer Shootout

The purpose of this subwoofer shootout is simple: compare products for performance using measures and techniques as defined by the CEA 2010 standard. The CEA 2010 standard is a means for measuring the maximum output of subwoofers at the two lowest audible octaves, 16-32 & 32-63 Hz. Unlike a traditional harmonic distortion number, the CEA number attempts to weight different harmonic distortion components based not only on their amplitude relative to the fundamental, but on their separation in frequency as well.

Example of Product with Excessive 3rd Harmonic Distortion
Fundamental Frequency =60 HZ, 2ND = 120 HZ, 3RD = 180 HZ, 4TH = 240 HZ, 5TH = 300 HZ, ETC.

Above is a spectral plot of the output of a 12" subwoofer in a box tuned to 40 Hz. While this author found this level of third harmonic distortion easily perceptible, I must note that this output would be considered well within the CEA third harmonic distortion limits imposed by the CEA standard. One of those individuals who had taken part in the creation of the standard has personally voiced opinions to me that perhaps the CEA standard is too lenient. I have also heard this comment from some of the vendors who submitted product for our own testing. Those vendors, such as SVSound, who held themselves to a higher standard than the CEA published "acceptable" distortion limits, actually placed themselves at a disadvantage because they had the system "draw the maximum line" at a location lower in distortion than CEA allowed for what it considers to be acceptable at maximum usable output. One must bear in mind, that both an amplifier with a built in compressor, as well as a loudspeaker, do not have a brick wall limitation in performance. (While technically possible to build an amplifier this way, the resultant distortion it would produce when it hit the threshold would be egregious. Some of you already know it, and refer to this as “Hard Clipping”.) When an amplifier is rated on power, the number is meaningless if not accompanied by a distortion figure, and whether or not the output is peak or RMS (which most of us think of as continuous instead of transient or short term.) The same is true when testing subwoofers for maximum output. You might be able to drive them past their intended maximum limit, but at what price in distortion? The CEA standard not only attempts to define that limit, but does so in a completely quantifiable way. Note the graphic below titled Signal Spectrum.

20 Hz Fundamental output within CEA allowable distortion limits

Please note the stepped red boundary which looks a bit like the side-view of a staircase. That red line represents the CEA 2010 Sub-Woofer standard limit on distortion. It does not represent a certain amount of distortion in absolute terms. It does shows the acceptable upper limits of distortion as a percentage of the fundamental output. (The test frequency). Note the center of the large “finger” shape in blue. It is centered at 20 Hz, in this case the test frequency. Notice a second finger centered at exactly twice the frequency, 40 Hz. That is second harmonic distortion. For this product, that is the dominant distortion when driven at 20 Hz to this output level. (It may well change if driven harder towards its physical limitations). Note also, the level of 0dB at the peak output centered at 20 Hz. This is NOT a measurement of the output in dB relative to any particular sound pressure. The program captures the signal output, and transforms the signal into a spectrum. Whatever the sound pressure is at the driving frequency (in this case 20 Hz), this peak value will be considered the reference, or “0 dB” value. It means that the other frequencies present are measured RELATIVE to the fundamental, which is why distortion can be expressed as a percentage. Notice that as the harmonic frequency is farther and farther away from the driving frequency, the CEA limitation, (like your ear) is more and more intolerant. The red stepped line means that the second harmonic distortion can be up to 10 dB less than the fundamental, the third an be up to 15 dB less, the fourth and fifth 20 dB less, and so on. This is weighting the distortion spectrum in a way that corresponds to how the human ear-brain mechanism responds. Since the distortion components are all below the threshold of the stepped red line, (CEA standard limits) this speaker at this test volume passed.
During my testing, I was sitting about 40 feet away (to the side) of the units under test. While I cannot determine how different the spectrum of sound was at the microphone relative to my position, on a relatively small number of occasions when the software recorded the output as a "FAIL", or "PASS", I might have decided from listening alone, that should not be the case. Despite that, I would argue that the CEA 2010 standard is as good or better than any other metric I can name for relating subwoofer distortion to usable output. The vast majority of the time, my ears would draw the line at the same levels the software would report a "FAIL". Let's take a look at the graphics shown to the test operator in the image below which shows a failure due to an excess of third harmonic distortion (60 Hz resulting from a 20 Hz input).

Excess Third Harmonic Distortion as per CEA 2010 Standard

As we can see from this example, the distortion of this speaker under test is predominated by third harmonic distortion (60Hz), and it rises in excess of what the Red stepped curve would allow as a maximum limit. At this point, we have crossed the line (literally) and are “in the red”, so the SPL produced by this speaker was not considered “usable”. That is, in a nutshell, how this program works. It produces ratings of Pass or Fail only. The only two discernible frequency components in this upper graphic are the fundamental driving frequency (test pulse centered at 20 Hz) and the 3rd harmonic distortion, centered at 60 Hz. During testing, the frequency center is changed from 20 to 63 Hz in 1/3rd octave steps (20, 25, 32, 40, 50 & 63 Hz). Unlike a traditional tone burst, which consists of a single frequency, this stimulus has a 1/3rd octave width. Traditionally 1/3rd octave is considered to be “critical bandwidth” of the human ear, meaning we lack the ability to distinguish tones that are within this close to one another in frequency. What does that mean? Well, take a look at the first graphic (#1). You can take note of the single vertical blue line which looks like it is centered at or about 550Hz. This is a very narrow-band noise, which likely has less to do with the speaker under test than it does to noise being emitted by a nearby factory. This line is how a traditional tone burst might appear, very narrow in comparison to the 1/3rd octave pulses used in testing. The advantage of 1/3rd octave signal pulse for testing, is that it will tend to average a speakers strengths and weaknesses over that 1/3rd octave. It is ENTIRELY possible for a speaker to have a big HUMP centered at EXACTLY 40 Hz, while at 35 Hz, it might have a huge and narrow notch. A traditional 40 Hz tone burst would show a huge output yet would be blind to the notch. Since 35 Hz is not a standard frequency center, this defect might go unnoticed by tone burst testing stepped in a similar way, 1/3rd octave at a time. With the 6.5 cycle burst from Don Keele's CEA test software, running on the graphics program Igor Pro, the nature of the signal is to be DOWN -3dB at a 1/6th octave increment on either side of the center frequency. (Meaning these pulses are 1/3rd octave wide). This is in fact quite similar (though not identical) to the kind of signal proposed by Siegfried Linkwitz to be used for testing on his site.
(http://www.linkwitzlab.com/sys_test.htm Scroll down to the section titled shaped tone burst generator).
Why do we care? Is this just to make my job easier? (As if anyone really cares!) No, this is important as it is a useful way for us to collect data in a way that corresponds to how we hear, without having thousands of data points to represent all the single frequencies we can perceive. It is also exciting the speaker with a signal far closer to music, as instruments tend to make sounds of a complex nature, containing many frequencies simultaneously. It is common to test a speaker with a single frequency, either stepping or sweeping it up and down to “connect the dots”, yet this kind of excitation is essentially unheard of in music. Musical tones are full of harmonics and generally have a wide spectrum looking completely different from traditional test tones.

The Ground Plane Testing Method
All of the subwoofers in the review were placed on the ground, in a large parking lot free from obstructions or objects from which sound would be reflected save for the ground.

While this is the method shown in the CEA standard, I imposed my own "peculiar" take on this based on 20 plus years of experience in measuring vented sub-woofers in real world rooms. This method as shown will emphasize the output of the vent relative to the driver output because of proximity errors, (in other words the mike is closer to the port than it is to the speaker). Another source of error will occur as differences in the size of the speaker vs the port are common. (Because the port is smaller, it contributes more at very close distance from the mike than the larger surface area of the sub, even if far far away, their contribution would be equal). My solution was simple, if not perfect. I turned the box on its side, and measured the microphone relative to a line midway between the center of the ports, and the center of the driver. This would tend to give me greater accuracy than the method as pictured above, and all subwoofers with this configuration were tested on their sides for this reason. This is something we hope CEA will notate in their next revision of the standard.

Subwoofer Box orientation with ports and driver both against the ground

This is how I tested the Rythmik sub in my parking lot. Notice the ports are to the side of the driver, not underneath as you would place it at home.

The area where the subwoofer was placed was on a carpet to protect the cabinets and microphone, on the ground as per the guidelines shown in the CEA 2010 standard. This mike was placed on the ground 39.37 inches (that's a meter to you mate) from the front of the baffle (no grilles were used for front firing woofers). This technique, while reasonable does place larger woofers, and systems with side firing woofers and/or ports at a disadvantage in large part because it still requires the meter be measured to the front of the cabinet instead of the centerline between speaker and ports or passive. In my opinion, the standard groundplane measurement as we previously illustrated deals with those configurations in an unfair way.

This may be in large part due to many years of precedent in calling for measurements to be made relative to the front of cabinets instead of the acoustic center of the radiator(s). That is a simplification which is reasonable, even if an additional source of error.

CEA 2010 Proposed Alternative Groundplane Measurement Technique for Alternate Driver Port Configurations

This tradition of measuring the SPL at 1 meter from the cabinet face (or side) may be in large part due to many years of precedent in calling for measurements to be made relative to the front of cabinets instead of the acoustic center of the radiator(s). That is a simplification which is reasonable, even if an additional source of error. (One could point out that a deep woofer actually mounted to the front panel, has an acoustic center farther away than the shallow woofer mounted in the same box. At some point, we have to pick a convention and stick with it, regardless of its imperfection.) One way to make this issue less bothersome is by increasing the measurement distance to 2 or 3 meters. The problem with that solution is that the further away from the speaker you get, the lower the ratio of direct sound versus the reflected sound which is essentially a function of the acoustic space in which you measure, and not the speaker itself.

Testing subwoofers accurately and without the response of the room added to your measurement requires either an outdoor measurement setup, or a very very large anechoic chamber, the kind of which few in the world exist large enough to be usable down to the lowest frequencies we can hear (16Hz). I being a poor soul, devoid of either a chamber or a 90 ft pole with a platform and ladder, allowing me to do true “free air” testing, opted for the outdoor method. It is common knowledge among those of us who measure sound frequently in real rooms that once your mike is more than a cm or two away from the cone, most of what you are measuring is room reflection, not speaker output. So, how do we get around this problem without a multi-million dollar room that will eat all reflections? (An anechoic room). We measure outside. But what about ground reflections? Are they not also a source of errors? Yes, unless you place the microphone directly on the ground so the “reflection” is actually in phase with the direct sound! This method is known in the industry as “ground plane” measurements.

Is this the best way to measure the speaker? Shouldn't it be up in the air, pointing directly at the center of the subwoofer? Those are all good questions, so let's take a minute to consider how sound forms around the speaker box at very low frequencies.

There is a school of thought that says, ground-plane measurements are too, a source of less than perfect information, so they prefer to climb a tall pole and hoist their speakers far above the ground. On a clear, sunny and windless day, this is indeed, the best of all methods (minus that huge anechoic chamber) but this particular writer has another peculiarity, fear of great heights. So, between that and the CEA standard as cover, I opted for the ground-plane method. For those unfamiliar with the concept, let me explain it like this. First of all, we need remember what sound is, a wave motion propagated through particles (air molecules) which end up in tightly or sparsely populated groups with densities greater or lesser than ambient air pressure. The louder the noise, the more intense the difference between the maximum and minimum density of air molecules.

Softer sounds barely effect the relative pressure at all. Since the speed of sound is a relative constant (it varies slightly with temperature pressure and humidity), we can calculate the distance between the pressure peaks and minima's in a waveform based on the frequency. The relationship is simple and stated thusly: $C = \lambda * F$ (Speed of sound (C) = Wavelength (λ) times frequency (F)). Since C is a constant, if we know the frequency (F) we can calculate λ (lambda) which is the distance between the peak-to-peak pressure (either negative to negative or positive to positive) in whatever units the speed of sound is expressed in.

2 Full Cycle of Sine Wave – Single Frequency vs Time

Let's do some simple math now. I want to know how long a 20 Hz wavelength is. In the graphic above, if the horizontal time scale is equal to 0.100 seconds, and I have two cycles as shown, my frequency would be = (2/0.100 seconds) = 20 Hz. My normal C is 343 Meters per second = 13,504 inches per second = 1125.3 ft per second. So expressed in inches, the wavelength at 20 Hz is 13,504/20 = 675.2 inches long, or about 56.27 ft or 17.15 Meters long. Wavelength is the distance that 20 Hz covers in one cycle and it takes 50 milliseconds to complete. Another way to say the same thing is the period of 20 Hz is 50 milliseconds. (Period = 1/F) At 200 Hz, the wavelength is 1/10th as long, and at 2000 Hz, the wavelength is 1/100th as long; and at

20,000 Hz, the wavelength is exactly 1/1000th as long or only 0.675 inches. Remember, since the speed of sound is essentially constant, what changes as we change the frequency at which we excite the air, is the number of peak and minimum pressures within a given distance. We call one cycle of distance the wave-length (Length of the wave, of course!)

The 6.5 Cycle Waveform Used In Don Keele's CEA Testing Program

Above is the graphic which shows the waveform used in the CEA test program. This waveform has the requisite 1/3rd octave width needed for testing. An important note regarding this test is that the program, and the numbers which we will publish in the specific product reviews are all representing the PEAK output of the subwoofer, NOT the RMS (root mean square) value which is so often the gold standard used to quantify amplifier power, or loudspeaker power handling. This is appropriate because a sine wave has only a 3dB peak to rms ratio. (Meaning a 100 watt RMS sinewave, has a peak value of 200 watts). Subwoofers have such a restricted bandwidth, only the least expensive of them ever get destroyed by heat or rms power delivered. The real test for a subwoofer is in the amount of peak power it can take, and more importantly the peak SPL within reasonable distortion limits it can deliver.

For the CEA measurements, while the harmonics are measured up high, the highest fundamental frequency of importance is 80 Hz, where the wavelength is 168.8 inches long. Because this wave-length is still large compared to the size of the box, and the distance from the acoustic center to the ground, our measurement error resulting from ground-plane measurements remains small. (Of course, since we are looking up to the sixth harmonic of the maximum driving frequency center (63 Hz), we might also want to know how big that 378 Hz wavelength is, since we are not on the speakers main axis, and judging its output up that high by looking at the distortion components!) As the frequency increases, and the box and speaker are no longer small by comparison, the sound no longer envelopes the enclosure or propagates evenly in every direction. At a high enough frequency, the errors from the ground-plane method become so large as to make the measurements useless. That is not the case for the subwoofer bandwidth. Here, the wavelengths are large enveloping bubbles of air that move about equally in all directions. Unless of course, you are on the ground. Here, the air stops, and the vibrations move out not as spheres, but hemispheres. (See graphic below)

What this means is a number of things. If the cabinet is small compared to this wavelength, then placing it and the microphone both on the floor is a pretty good way for us to approximate with relative ease what the speaker does at LOW frequencies ONLY, provided we have a big enough parking lot, and a long enough mike cable, and the sound from the speaker is loud relative to the background noise you are subject to.

Ground Plane Measurement

This relatively useful approximation to true "free space" radiation (sometimes called 4 pi) is called half-space or (2-pi) radiation. In our case it's technically more correct to refer to it as groundplane since the driver is not flush mounted to the actual ground. In simpler terms, the bubble forming about the speaker (which is at the center of the radiation) is forming a hemisphere about itself, not a complete sphere. If the size of the wave-length were near or less than the distance from the acoustic center of the port or speaker, (relatively high frequencies) this approximation would be poor, and the ground plane technique would introduce intolerable errors. At very low frequencies however, it remains both a practical and well accepted practice among sound professionals for characterizing very low frequency performance of loudspeakers. You will notice the ground-plane technique produces not a full sphere, but rather a half (hemi) sphere of sound. Since we are maintaining the same pressure in half the volume, we find our ground plane measurements show an increase (approaching, yet never exceeding) 6dB compared to the true free space 4 pi measurements (measured at the same distance). Those are the numbers we will show here, and while this results in a source having an effective height double to that made with a free space measurement, the error is acceptably small. (Unless you went out and bought a 90 ft pole, and a really tall ladder already.)

Mirror Image Equivalent of Ground Plane if Done In Free Space

You can think of ground-plane technique like a free space technique where you have two subwoofers stacked one on top of another. The ground itself acts a bit like an acoustic mirror, reflecting the wave that hits it.
For more information about 4pi, 2pi and groundplane, please read: Subwoofer Measurement Tactics

Test Gear Used

- The microphone used was a brand new Earthworks M30 omnidirectional laboratory grade measurement microphone (http://www.earthworksaudio.com/our-microphones/m-series/m30/) which came with a calibration.
- Microphone preamp = True Systems P-Solo (http://www.true-systems.com/p-solo.html)

- Echo Mia Midi Soundcard (http://www.echoaudio.com/Products/PCI/MiaMIDI/index.php)
- Dell Optiplex Computer running Windows XP SP3
- Don Keele's CEA 2010 Measurement software (God bless Don Keele!)
- Lots and lots of cables of every kind, mostly 1/4" TRS and/or XLR Canon
- Far too many large diameter extension cords
- B52 Professional Audio's parking lot (when the rain finally let up)

CEA 2010 Subwoofer Definition

Right off the bat, I was less than impressed with this standard when I got to section 2.2, where they define subwoofer as "A speaker designed to reproduce all or a portion of the audio signals below 120 Hertz (Hz)." Well, by that definition, a mini speaker I use next to my PC which has no bass to speak of, yet a corner frequency of 100Hz (-3dB pt.) is a subwoofer. Clearly this revision of the standard has not yet been put under a lot of scrutiny, but while on the subject, let me attempt to define subwoofer, a word which I am frequently reminded by my spell-checker does not exist as I write it.

A subwoofer is a speaker which is designed to specifically reproduce the lowest usable frequency range of any given sound system. It is usually characterized by a relatively large size of both the cabinet and driver, and generally has limited usable frequency response above 200Hz, especially off the main axis. The term Sub-Woofer (meaning literally below the woofer) is an indication of the intent of its use below (frequency band) a regular woofer, which typically has a frequency range sufficient in bandwidth to cross over to a mid range or high frequency loudspeaker. How is it we all know that, yet the CEA standard seems to define so poorly the very thing which it displays expertise at measuring? I will ask them, and see if they will agree to a different definition.

What's Next?

The actual system testing of course. We are going to bring the subwoofers outside, and find their maximum output by virtue of the CEA testing standard we touched on above. The following articles will have more detail about the actual levels you can expect, as well as some simplified physics, so you can all become experts at judging subwoofers in your own right.

Subwoofer Measurement Tactics: A Brief, Topical Overview & Method Comparison

by Mark Sanfilipo — November 05, 2007
Contributors: Gene DellaSala

Axiom Sub

The challenges of accurately capturing the direct sound amplitude response of a subwoofer, especially for the enthusiast who would not likely have access to the facilities or hardware typically available to the professional, are well known and widely discussed in print and online. In this brief overview, we'll take a look at a variety of measurement approaches that when correctly implemented can minimize or in some cases altogether eliminate the complicating influence of room\boundary\subwoofer interaction. The scope of this overview is limited to subwoofers only and the amplitude response frequencies range of 10 Hz to 320 Hz.

This is by no means a detailed treatise of the subject. For a more in-depth treatment of this topic the reader is encouraged to read through the resources listed in the bibliography.

Measurement Approach/Domain Space	Implementation	Advantages	Disadvantages	Limits
Anechoic Chamber	Acoustical measurements done within an indoor, (ideally) reflection-free environment	Climate-controlled, artificial environment in which to measure amplitude response, noise & distortion, diffraction effect & directional response characteristics	Cost; extremely large chamber needed for accurate LF amplitude response, noise & distortion, etc measurement	Chamber , device under test (DUT) size; depth, type & configuration of absorptive material used within the chamber

Tower/Crane Measurement	Acoustical measurement done out of doors, with DUT mounted on a tower or suspended by crane	Under appropriate conditions, can provide for an ideal free-field measurement environment	Cost, Requires DUT be placed well away from any reflective surfaces or objects large enough to influence measurements	Noise pollution, inclement weather
Ground Plane Measurement	Measurement done with the DUT & microphone typically placed on the ground, with the emissive radiating surface(s) pointed at the microphone	Low cost; ease of implementation, within known limits, can provide accurate measurement data	Upper & lower frequency limits. Other than the ground upon which it rests, DUT must be placed well away from any reflective surfaces or objects large enough to influence measured amplitude response	Noise pollution, Inclement weather (when done outdoors)
Half-Space or Hemispherical Free Field Measurement	Device affixed flush-mount with surface such as baffle, ground surface or clear wall of Hemi-anechoic chamber	Depending on implementation and type of data sought, can provide excellent results	Cost of indoor Hemi-anechoic chamber; use of baffle invites cancellation?. Out door, in-ground placement requires DUT to be placed well away from any reflective surfaces or objects large enough to influence measurements	Approach requires all emissive radiators be on one side of the cabinet
Windowing	Measurement taken and unwanted data windowed out	Fast data acquisition & post-processing	Requires significant data post-processing and the ability to skillfully interpret the results	LF measurement accuracy defined by environmentally determined window length. Poor tolerance for time variance.
Near Field Measurement	Measurement done with microphone placed near to, centered on and	When implemented correctly, the near-field amplitude	Multiple emissive surfaces require multiple measurements along with	Upper frequency limit determined by size of DUT.

normal to front emissive surface of each acoustic radiator	response provides for an accurate facsimile of the far field response	subsequent post processing

I. Anechoic Chamber

An anechoic chamber, in the literal sense, is a “chamber without echo”. Generally speaking, a chamber is commonly considered anechoic at any particular frequency when 1% (or less) of the acoustic energy impinging on the absorptive material lining the walls of the chamber is returned to the clear space or “lumen” of the chamber. The idea here is to create a *free-field* acoustic environment in which to assess a variety of loudspeaker acoustical performance characteristics such as amplitude response, directionality, distortion and so forth.

The necessary free-field conditions occur when the sub appears as a singular or sole point source, radiating spherically (4π sr) into an anechoic environment. (The sub behaves as a point source when the wavelength, λ, of the sound radiated is >> than its largest dimension). The measurement mic is then positioned far enough away from the sub so that it is within that portion of the sub’s sound field where the sound pressure level drops -6dB for every doubling of distance. The mic would now be considered in the sub’s Far Field.

Being accustomed as we are to the ever-present noise of everyday life, stepping into the reflection-free silence of an anechoic chamber can be an odd, vaguely disorienting experience. Paradoxically, its precisely those characteristics that give rise to this odd sensory experience of that very artificial environment which make it ideal for the purpose of capturing accurate data needed to correctly assess various acoustical performance characteristics of a subwoofer.

Though there are certainly challenges to be found in designing a chamber to perform anechoically in the mid- and high-frequency portions of the audible spectrum, it’s the low frequency portion of the audible spectrum where practical limitations come into play and the design process becomes especially challenging. When the 1% reflected acoustic energy figure cited above is reached, that’s typically considered the useful (ie anechoic) LF limit of the chamber being assessed.

These LF performance limitations are defined by a number of factors, including the overall linear dimensions and enclosed volume of the chamber, the depth/distributed density of the absorptive wedges populating the interior surfaces of the chamber, (along with the mechanical/acoustical properties of the wedges themselves) and the size of the DUT. A chamber useful for measurements down to ~ 30 Hz would require an internal volume of over 4700 m^3 (166k ft^3) and sport wedges over ~3m (9.8 ft) in depth.

The basic principal behind the typical anechoic chamber design is that of a collection of boundaries, functioning as purely resistive acoustic loads, absorbing progressive plane waves. The assumption made here is that the absorptive surfaces are hit by the waves, which themselves sport a specific acoustic impedance, Z, which is purely real. This assumption holds true if the distance from the source (namely, your subwoofer) generating the spherically diverging waves to the absorptive wedges is sufficient in distance so as to present, for all intents and purposes, progressive plane waves to the boundaries. This assumption does not necessarily hold true at low frequencies, hence the LF limits.

At right are two amplitude response plots of the same sub. The blue is that of the sub measured outdoors, on a tower and raised 100' in the air. The red plot is that of the same sub, now measured in an anechoic chamber. The anechoic chamber's ability to absorb & dissipate acoustic energy in the < 30 Hz band progressively diminishes as the measurement frequency approaches 10 Hz. The chambers absorptive ability's have clearly been exceeded.

Increasing the linear dimensions of the chamber and the depth/density (that is to say the resistive/reactive characteristics) of the absorptive wedges is one way to extend the LF limit of a chamber. An alternative, typically less costly approach, is constructing absorptive surfaces such that they present a reactive boundary at low frequencies and a resistive boundary at mid- and high-frequencies. Referred to as an "acoustic jungle", the absorptive arrays, rather than the usual wedges, comprise arrays of absorptive blocks, differing in size & density, built up in a way such as to present just such a reactive/resistive boundary surface.

Practically speaking, making use of an anechoic chamber to measure your sub can present a variety of possibly insurmountable challenges to the audio enthusiast: rental cost, lack of access, extremely low SAF if you build one (unless you own the company). Nevertheless, from a strictly scientific or engineering standpoint, anechoic chambers remain an ideal, indoor measurement environment.

II. Outdoor (Tower/Crane) Measurement

For the pro & audio enthusiast alike, outdoor measurement is an attractive alternative. At its best, it can provide for a measurement environment that rivals (or in some cases) surpasses that of an anechoic chamber. The downside, of course, in taking your subwoofer & measurement gear outdoors is that you now have the weather as well as background noise pollution to contend with.

But let's suppose for a moment the weather is cooperating and the background noise pollution is minimal or otherwise of a nature that a dose of curve-averaging can properly deal with. How then to attain at or near anechoic measurement conditions?

One method is to mount the sub on a stand, pole, tower, boom placed well above the ground or haul it up with a crane. There are, of course, always practical constraints (as well as safety issues) involved where it comes to just how high above the ground you can place the sub. Strictly from a measurement perspective, though, the higher the better.

The idea here is to get the sub as far away from any response-muddling reflective boundaries as possible. Given the wavelengths involved (a 20 Hz wave is approximately 17 meters (56.5

feet) long) it can be quite challenging, if not altogether *impossible*, to get the sub as high above the ground as you might like. Nevertheless, get the sub high enough and you will be measuring in an ideal anechoic, free field environment. Indeed, data taken under such conditions can be clean & accurate enough that you can build mic correction files with it. Because the quality of measurement data taken under the aforementioned conditions is so good, the baseline amplitude response used for comparison purposes in this article will be that of the test sub measured at the top of the tower. This correction file could be used to develop an accurate low frequency response measurement in an anechoic chamber.

If you don't own or otherwise have access to a 100' tower, like that showing in Figs. 3a & b above, or any other means by which to remove the sub from the vicinity of any reflective surface, no need to worry; there are still other approaches that can be used that in practice cost little or nothing.

Plot 1: Akabak Model: 12" Driver in totally enclosed box. Note how curves progressively flatten as height is increased.

The next best alternative to measuring a subwoofer on a pole would be to place it on the ground. This measurement is known as the "Groundplane technique" which is our next topic of discussion.

III. Ground Plane Measurement

We now come to an option that doesn't require much more than a flat, solid, unobstructed surface such as an asphalt or concrete driveway or parking lot to rest the sub and measurement microphone on. Obviously, in being outdoors, you'll once again be at the mercy of the elements as well as background noise pollution.

Don't underestimate the latter's measurement-corrupting capabilities: air or ground traffic or wind noise can easily mangle an otherwise good amplitude response measurement, as evident in the < 15 Hz portion of the blue and < 25 Hz portion of the red curves seen above . Depending on the nature of the background noise, redoing the measurements when all is quiet or averaging several measurements are effective antidotes.

Where the tower approach represents an attempt to eliminate boundary reflection by placing the sub well away from any reflective surfaces, ground plane measurement takes into account the effect of the single boundary the sub sits on. The acoustic signal reflected from the ground surface is considered to be a second, *virtual* source, identical to the actual source.
Based on the well proven, time tested method of images, ground plane measurement is an approach solidly grounded in science. To see why it works as well as it does, let's look at the equation for the magnitude of the rms sound pressure, $|p|$:

(1)

Where:
$|p|$ = magnitude of the rms sound pressure
A = magnitude of the rms sound pressure at unit distance from the center of each source
r = measurement distance
b = distance between the center of actual and virtual acoustic image
λ = wavelength of frequency under consideration
θ = angle to the perpendicular bisecting the actual and virtual acoustic images

Note that eq. 1 is for *2* acoustical sources, in this case the sub's actual and virtual acoustic images.

At low frequencies (ie long wavelengths, where $\lambda >> b$), b is relatively small by comparison and the actual & virtual source will appear as a *single* source with sound pressure *double* that of the actual source alone. This doubling results in a 6dB increase of the measured axial sound pressure level above that produced by the same sub measured at the same distance under perfectly anechoic, free field conditions.
Care must be taken to ensure that measurements are, ideally, made with no large objects or reflective surfaces (other than the ground the sub sits on) within a radius $\geq \lambda_{LF}/2$, where λ_{LF} equals the wavelength of the lowest frequency of interest. In this case a large "object" might be a barn or your neighbor's house; a large reflective surface might be, for example, the side of a nearby apartment building or office tower.

Plot 2: Akabak Model: 12” Driver in totally enclosed box. Note interference effects from nearby boundaries.

Setting up for a ground plane measurement is pretty straightforward. Position the sub so its driver(s) are facing the measurement microphone; that is to say, lay the thing on its side. Then tilt the cabinet until the center axis, normal to the driver (or panel into which the drivers have been bolted, if 2 or more drivers are used) points to a spot on the ground 2 meters away. You may find using a laser pointer handy. Once the subwoofer has been oriented correctly, place the measurement microphone at the exact spot where the axis intersects the ground and measure away. But why measure at 2 meters when amplitude response measurements are typically made at 1 meter?

When the virtual & actual acoustic image outputs combine at the microphone, the net effect is to add 6dB to the axial sound pressure level. Doubling the measurement distance from 1 meter to 2 meters decreases the axial sound pressure level by 6 dB. As mentioned above, do so and the net result at the frequencies of interest produced by the subwoofer, is that the response will *very* nearly mirror that of the subwoofer measured free-field (or in an ideal anechoic chamber) only *slightly* altered by a few unavoidable acoustical artifacts.
Working up a model (15” driver in totally enclosed box) in LEAP 5 (plots at right), we see the reference free-air curve in blue, ground plane @ 1 m in purple and the ground plane curve scaled to 2m (red). As expected, the ground-plane at 2m is nearly identical to the free-air plot at 1m. Clearly, the ground-plane approach is an excellent, cost-effective (if not outright free) alternative if you don’t happen to have handy an anechoic chamber or a 100’ tower.
Given the ease of implementation, the data quality typically attainable and the minimal post-processing requirements, the ground-plane measurement technique (when done outdoors) is likely the best options available to the audio enthusiast.

IV. Half-Space or Hemispherical Free Field Measurement

This approach places either a raw driver or the front panel of the loudspeaker system within and flush with a large baffle. The rationale often cited for making use of this approach is that in changing the domain space from 4π steradian (spherical) to 2π steradian (hemispherical) the resulting amplitude response measurements will more accurately portray the low frequency performance of the system within a typical listening environment.
In practice, the usual implementations of this approach make use of either a simple, large baffle, (as per IEC standard 268 -14) for driver testing, a hemi-anechoic chamber or simply burying the

system in the ground (or other large flat surface, such as a roof) with the driver or the faceplate of a complete system positioned upward and flush with the test surface.
From a practical standpoint, digging a hole in the backyard to measure your sub's amplitude response isn't typically found in the list of items guaranteed to result in a positive SAF. Besides all that, what if your sub is a vented system with the ducts firing out any panel *other* than the one the driver(s) are bolted into - how are you going to bury that?
Modeling again in LEAP 5, the plots showing at above left are free-air (blue), half-space @ 2m (red) and half-space @ 1m (purple).

V. Time-Windowed Impulse/MLS

The time-windowed measurement approach has a number of immediately attractive qualities that have made it a popular tool when it comes time to assessing various acoustical performance characteristics of a loudspeaker. Essentially, it works as follows: an impulse or *M*aximum *L*ength *S*equence signal (MLS: a deterministic signal with spectral properties similar to that of white noise) is first applied to a sub and the measured response is then run through a variety of post-processing utilities (See figure, above, Left), in turn producing ETC curves, frequency response plots, cumulative spectral decay (aka waterfall) plots and so forth (See figure, above, right).
Probably the most intuitively appealing feature of this particular approach is the option to window out all but the anechoic portion of the measured acoustical signal. Sounds ideal for subwoofer measurement, doesn't it? However, in practice this approach is limited in terms of how accurate the LF results can be by, among other things, the prevailing acoustical/environmental conditions.
Suppose for a moment you're measuring your sub's performance indoors and owing to reflections you find you need to window out all data beyond 10ms. Within that time window, the largest complete sinusoidal period that can be completely contained is 10 ms long. Any data featuring wavelengths possessing a longer period than that (i.e. anything lower in frequency than 1/10ms = 100 Hz) will be inaccurately presented, more so the lower the frequency. Any inaccuracy existing in the initial, raw measurement data will of course carry through to whatever results are generated through post processing.
As dreary as this might sound to anyone contemplating using this approach in measuring a sub, there are workarounds that allow for accurate time-windowed measurements - even under less than ideal circumstances - such as preprocessing the test signal, measuring with a mic specifically calibrated for the environment/set up your using or making time-windowed measurements in the near-field, a topic covered in the next section.

VI. Near Field Measurement
The near field approach, when done correctly is a simple, yet *very* effective means for capturing the direct sound amplitude response of your subwoofer. Where it comes to measuring subs, many of the practical limits or issues faced when employing some of the alternate approaches illustrated above are simply rendered moot at the frequencies of interest. If your only option is to measure your subwoofer indoors, this is probably the best – and easiest - approach to take.

From Eq 2:

(2)

Where:

p_n = peak pressure in the near field at the center of the piston (driver diaphragm)

r = distance from measuring point (mic position) to the center of the piston

a = piston radius

, peak axial pressure measured at a distance, r, in the far field of the piston

ρ_o = density of air, = 1.21 kg/m^3 at 20°C

U_0 = piston peak output volume velocity

k = wave number, = $2\pi/\lambda = \omega/c$

c= velocity of sound in air, = 343 m/s

We see that for those frequencies of interest (ka < 1), the near field sound pressure is directly proportional to far field sound pressure and the measurements made in the near field are basically *independent* of the space into which your subwoofer is firing; measuring "near" and scaling to "far" is valid & accurate. The upper frequency boundary for accurate near-field measurements is reached when a/λ ~ .26

In addition to being an effective means by which to measure a sub's amplitude response, this technique (when done correctly) also allows for equally valid measurements of system distortion, efficiency and total acoustic power. It can be used to measure either closed or vented systems, powered or passive.

The near-field sound pressure measurement technique requires the measurement microphone be placed in a position centered on and normal to the driver's dustcap and no further away from the center of the radiator than .11a (r ≤ .11a), assuming measurement data accurate to within 1dB or less is the goal. If you're measuring a vented system, you'll need to measure separately the acoustic output of the driver(s) and port(s). When measuring the latter, you'll need to place the mic centered on & normal to the duct's port, flush with the system's faceplate.

Where there is only one radiator, such as a sub comprising a single radiator (driver) in a totally enclosed box, you need only measure the driver's near-field output and scale the resulting data. If you're measuring a vented system (or any system featuring more than one radiator, be it port(s), driver(s) or passive radiator(s)), then individual measurements are made of each radiator and the data are then combined to produce the total system amplitude response.

Because all measurements are being taken near-field, the data should be scaled to an appropriate distance, commonly 1 or 2 meters. So how do you scale the nearfield measurement data to, say, 1m? To calculate the near-field scaling factor (1m, half-space) use:

$$FF_{dB} = NF_{dB} - 20 * Log(0.2821 * SQRT(Sd)) \text{ (dB)} \quad (3)$$

Where: Sd is the effective diameter of the radiator, m^2

(To calculate the 1m, *full-space (anechoic)*, far-field equivalent, use the above equation and subtract 6dB from the results).

Here are some alternate equations useful in scaling NF to FF measurement data:

$$FF_{dB} = NF_{dB} - 20 * Log(4d/r)\ dB,\ 4\pi\text{-space} \quad (4a)$$

and

$$FF_{dB} = NF_{dB} - 20 * Log(2d/r)\ dB,\ 2\pi\text{-space} \quad (4b)$$

Where:
FFdB = scaled far-field dB value
NFdB = near-field dB value
d = distance at which far-field values are to be calculated (eg: 1m)
r = effective radius of radiator
Note: units (m, in, cm, etc) must be the same for both d & r
Refer to the chart below for the Sd values for a variety of nominal driver sizes:

Nominal Driver Size (Diameter) Sd (m^2)	**Sd (m^2)**
24 Inch (610 mm)	0.2200
18 Inch (460 mm)	0.1300
15 Inch (380 mm)	0.0890
12 Inch (300 mm)	0.0530
10 Inch (250 mm)	0.0330
8 Inch (200 mm)	0.0220
6½ Inch (170 mm)	0.0165
6 Inch (150 mm)	0.0125
5¼ Inch (140 mm)	0.0089
4½ Inch (110 mm)	0.0055
3 Inch (80 mm)	0.0038

Should the distance between driver(s) and port(s) be significant in terms of the wavelengths of interest, the maths become a bit more complex if data of the highest quality is your requirement. Neville Thiele wrote a concise paper on the maths involved, referred to in the bibliography.
Before commencing any near-field measurements, be certain your measurement mic can handle the acoustic output dished out by the radiator(s) at close proximity. This holds especially true if you're interested in doing max. dB spl output testing.
Also, when setting drive levels its best to do a number of test runs, working up each time to determine driver excursion so you don't end up with the mic getting hit by the unit's diaphragm when doing the actual measurements. As well, to ensure accurate port data, avoid using drive levels so high as to introduce turbulence in the port.

Conclusion

In this brief overview we've looked at a variety of approaches to subwoofer measurement. Each has its own set of strengths, weaknesses and limitations. Success in capturing accurate data depends on correctly understanding how to measure, how to interpret the results and above all remaining cognizant of the limitations inherent to whichever approach is used. And if all else fails, there's not much a little 1/3rd-octave smoothing won't make look good.
As a handy reference, a graph featuring plots of a system modeled in the half-space, ground plane and free-air space domains, along with an accompanying table are included below.

	Relative Gain @ 1m, Drv. Level Held Constant	Comments
Free-space (blue trace)	0 dB (reference)	Half-space & Ground Plane amplitude response plots referred to this plot
Infinite Baffle (brown trace)	~+6dB @ low frequencies	Gain relative to free-space amplitude response varies with frequency. Diffraction plays no roll in determining amplitude response.
Ground Plane (green trace)	+6dB	Holding drive voltage constant, but measuring @ 2m gives results virtually identical to those obtained measuring Free-space

Resources

1. Allison, R. F. and Berkovitz, R.: “The Sound Field in Home Listening Rooms” Journal of the Audio Engineering Society, Vol. 20, July/Aug. 1972
2. Benson, J E., “Theory & Design of Loudspeaker Enclosures”, Howard W. Sams & Co., Indianapolis, IN, 1996
3. Berman, J. M. & Fincham, L. R.: “The Application of Digital Techniques to the Measurement of Loudspeakers” Journal of the Audio Engineering Society, Vol. 25, June 1977
4. Bjor, Ole H.: “Maximum Length Sequence” Norsonic AS, 2000
5. Brüel, Dr. Per V., “Anechoic Chambers”, Technical Review 96-02, Brüel Acoustics, 1996
6. D’Appolito, J.: “Testing Loudspeakers”, Audio Amateur Press, Peterborough, NH, 1998
7. Dunn, Chris & Hawksford, Malcolm., “Distortion Immunity of MLS-Derived Impulse Response Measurements”, Journal of the Audio Engineering Society, Vol. 41, #5, May 1993
8. Fincham, L. R.:, “Refinements in the Impulse Testing of Loudspeakers”, Journal of the Audio Engineering Society, Vol. 33, #3, October 1983
9. Fincham, L. R.:, “Production Testing of Loudspeakers Using Digital Techniques”, Journal of the Audio Engineering Society, Vol. 27, #12, December 1979
10. Gander, Mark R., “Ground-Plane Acoustic Measurement of Loudspeaker Systems”, Journal of the Audio Engineering Society, Vol. 30, #10, October 1982
11. Geddes, Earl R., “On Sound Radiation from Ported Enclosures”, Journal of the Audio Engineering Society, Vol. 49, #3, March 2001
12. Griesinger, David.: “Beyond MLS - Occupied hall measurement with FFT techniques ” Lexicon, Waltham, MA

13. Keele, Don B., “Anechoic Chamber Walls: Should They Be Resistive or Reactive at Low Frequencies”, Journal of the Audio Engineering Society, Vol. 42, #6, June 1994
14. Keele, Don B., “Low-Frequency Loudspeaker Assessment by Nearfield Sound-Pressure Measurement”, Journal of the Audio Engineering Society, Vol. 22, #3, April 1974
15. Klapman, S. J., “Interaction Impedance of a System of Circular Pistons”, Journal of the Audio Society of America, January 1940
16. Massarani, P & Muller, Sven.: “Transfer Function Measurement With Sweeps”
17. Olson, Harry F., “Acoustic Laboratory in the New RCA Laboratories”, Journal of the Acoustical Society of America, Vol. 15 October 1943
18. Oleson, Soren k. et. al.: “An Improved MLS Measurement System For Acquiring Room Impulse Responses”, Aalborg University, Department of Acoustics,
Aalborg, Denmark,
19. Pierce, R.: “Measuring Sensitivity & Power Ratings”, Speaker Builder Magazine, Audio Amateur Press, Peterborough, NH, May 1989
20. Preis, D.: “Impulse Testing and Peak Clipping”, Journal of the Audio Engineering Society, Vol. 250, #1/2, Jan/Feb 1977
21. Rife, Douglas D.: “Maximum-Length Sequence System Analyzer Reference Manual, v10WI, Rev 8” DRA Laboratories, USA, 2007
22. Strahm, Chris: “LMS Loudspeaker Measurement System”, Release 4.1, LinearX Systems Inc., USA, 2000
23. Thiele, A. N., “Estimating the Loudspeaker Response when the Vent Output is Delayed”, Journal of the Audio Engineering Society, Vol. 50, #3, May 2002
24. Thiele, A. N., “Loudspeakers in Vented Boxes”, Journal of the Audio Engineering Society, Vol. 19, May 1971
25. Various Authors, “AES Recommended Practice Specification of Loudspeaker Components Used in Professional Audio and Sound Reinforcement”, Audio Engineering Society, AES2-1984 (r2003) 1984