Format Politics • How Big Is Your Pipe? • HDR Marketing

The Essential Home Theatre Resource™

Widescreen Review®

7 Blu-ray Reviews

4 UHD Blu-ray Reviews

Phase Technology dARTS 660 Series

Marantz AV7703 11.2-Channel Processor

$5.99 US $7.99 Canada
Volume 26, Number 1
Issue 213, January 2017
WidescreenReview.com
Facebook.com/WidescreenReview

Widescreen Review®
www.WidescreenReview.com
Volume 26, Number 1, Issue 213, January 2017

Projector Placement And Why Vertical Projection Angles Matter

Dave Rodgers

There are various factors most installers know about when setting up a projection system, but the proverbial "devil in the details" often goes overlooked. I'm talking about a projector's vertical alignment. It is a relatively simple function, but it has small elements to it that every installer should be aware of. It may be a small matter, but the difference between doing it right and wrong will come with either great rewards or a potentially costly disappointment.

The Pitfalls Of Doing It Wrong

Keep in mind that calculating the wrong projection angle can create a multitude of problems for you.

• Keystone effect is created when the projected image has a trapezoid (trapezium) shape. This is caused when your projected light is hitting the screen surface from too high/low of a vertical angle. Most projectors have a keystone adjuster to correct this, but any projector without this feature should only be used in line with the screen. *Tip, make sure the projector you install can handle the angle of projection you want to achieve.

• Hot-spotting and glare is when a bright spot or glare appears in your projected image. It typically occurs with higher gain (more reflective) and specialized ambient light rejecting (ALR) projection surfaces. This is often the cause of off-axis projection or trying to use a short-throw (ST) projector with an ALR material. Albeit there are a few exceptions, most ALR materials are not designed for use with "ST" projectors. Specialty materials always present the greatest risk of occurrence, so it is best to check with the manufacturer if ever you have a doubt. *Tip, make sure that your screen material and projector are compatible with one another; they don't always get along with one another.

• Off-axis light loss is when a specialty reflective material, such as ALR, is used with either too steep of a projection angle or the entire screen and projector installation is not properly aligned with the audience's eye level. *Tip, matte white materials are best when you have good light control and the projector has good contrast performance. ALR materials are best when there is a lot of ambient light, but they come with their own limitations.

This article also serves to aid you in selecting your projection screen so that any alignment issues due to product incompatibility can be avoided. Factors such as projector type, atmospheric lighting, and screen material all come into play. Since your screen complements your projector, here are some basic rules regarding how to choose the right projector and screen combination.

Selecting the right projector is typically an easy task as long as you remember these guidelines:

• Get a projector that matches your screen's aspect ratio. Aspect ratio is the shape of the screen and projected image. Some aspect ratios are for a more squared image while others are of a wider format. For example, television sets of the 1950s to 1990s were typically in NTSC or (4:3). This means that for every four units of horizontal measure, there are three equal units of vertical measure. Later HDTV aspect ratios would be (16:9) or 16 units of horizontal measure to every 9 units of vertical measure. Whatever the case, it is important that your video content, projector, and screen are all of the same aspect ratio so that the image will be properly formatted.

• Make sure the projector's throw distance will produce an image that is the same size as your screen. There are numerous online calculators in addition to your owner's manual that will inform you on what size your image will be, depending on the "throw distance" or distance from your projector's lens to the projection surface. Most projectors also have a zoom function that will give you some leeway in formatting your image.

• Make sure that the projection angle will not conflict with the design of your projection material. White projection materials are the best for their wide diffusion uniformity. A projector can make a good image on it from either an in-line, high or low vertical axis. This makes it ideal for high ceiling or low tabletop presentations. Matte white screens are also available in the greatest range of sizes and aspect ratios. Specialty materials such as the new ambient light rejecting (ALR) are meant to be used with more in-line projection displays but have their own advantages as well.

Adjusting Your Vertical Angle

On a projector, there is typically a feature known as vertical lens shift. This is the projector's ability to shift an image upward or downward from a projector's static position without the need of physically moving the projector itself. Depending on whether you are ceiling-mounting your projector to shoot down on a screen from the front, or "table-top" positioning from in front or behind the screen, will determine the degree of "vertical adjustment" required from your projector. For high ceilings, a projector with up to three-times the screen height is required. For in-line imaging, typically from a table-top or low-ceiling installation, a projector that can only adjust up to one-half the screen height may likely be the ideal solution.

*Tip, there are times when you will be able to actually tilt the screen itself to accommodate the projection angle you have. This is often practiced with the angular-reflective "ALR" screen materials.

The Various Types Of Projection Materials And Recommended Angles Of Projection

Matte White

• Description: White front projection material

• Strength: Extremely versatile, wide viewing angle, widely available in small to large commercial sizes, can be used with either standard or short-throw projectors

• Weakness: Requires light control; will wash out from ambient light. Little if any contrast enhancement

• Suggested projection angle: In-line, high or low projection angle projection (see projector manual)

ALR, Angular Reflective

• Description: Ambient light rejecting, reflection angle mirrors projection angle

• Strength: Contrast enhancement, superior image brightness even with high levels of indirect lighting

• Weakness: Projection angles are limited, narrower viewing angle due to reduced half-gain cone, does not work with short-throw projectors

• Suggested projection angle: In-line with the possibility of some off-axis projection angle that will not exceed 15 degrees (*Note, tilting the screen may likely be required for optimal visibility.)

ALR, Retro Reflective

• Description: Ambient light rejecting, reflection angle returns directly to angle of projection source

• Strength: Top-tier contrast enhancement and excellent image brightness even with high levels of indirect lighting.

• Weakness: Projection angles are extremely limited and large venue sizes may not be available, slim visibility in vertical off-axis so high-installs are out, narrower viewing angle due to reduced half-gain cone, does not work with short-throw projectors

• Suggested projection angle: In-line projection only, preferably within 10 degrees

CLR, Ceiling Light Rejecting (aka. "ST" or "UST" Material)

• Description: ALR screen with specialized microstructures that specifically reject up to 95 percent of overhead lighting while moderately negating ambient light from other sources

• Strength: All the benefits of an ALR and is specially made for short throw projectors

• Weakness: Limited to smaller size options, standard projectors will not work with these

• Suggested projection angle: In-line projection only, never greater than 5 degrees

Rear Projection

• Description: Projector that facilitates images from behind

• Strength: opens up space normally restricted between projector and screen, superior brightness and some ALR properties

• Weakness: Typically does not function for front projection. May not be space conscious to your interior design, hot-spot artifacts are common

• Suggested projection angle: This varies by brand and model. Some are made for off-axis projection while others are exclusively in line—always consult manufacturer

Whatever your requirements are, always make sure that the desired "sweet spot" of optimal projector screen performance will encompass your entire viewing audience. Ambient light rejecting screens are great but they may not be as ideally suited to large room presentation as a white screen would be. Either way, always have a good handle on what your projector's vertical angle is and what angle of projection your screen can handle. If you do it right, you'll appreciate the results. **WSR**

Dave Rodgers is a guest writer for numerous publications and the Marketing Manager for Elite Screens Inc., with 20-years experience in the AV and wireless communications industries. He travels worldwide, playing an active role with international business development, and is regularly consulted in matters of branding and new business development. Dave has made numerous television, radio, and editorial appearances providing installers and Do-It-Yourselfers with easy solutions toward creating larger-than-life big screen applications.

Loudspeaker Accuracy

The Quest For Getting Loudspeaker Design Right

John Dunlavy

Editor's Preface

This series of articles appeared in *Widescreen Review* many years ago (March through September 2002), but they are just as pertinent today as they were years before. Loudspeaker design is both a science and an art. John Dunlavy, who passed away on January 4, 2007, was a master of both.

The articles are being presented together as one, in their entirety, exactly as they appeared in 2002's issues.

Introduction

I've never met a person who didn't love music. Even people who can't hum their way out of a paper bag get excited when describing their favorite "song"—whether it's classical, rock, bluegrass, jazz, or some other variation on a theme. And each one of them thinks his or her kind of music is the best.

When it comes to music, it's personal.

The more you care about music, the more you ought to care about getting it right. By right, I mean accurate. And by accurate, I mean acoustically identical to the performance, whether recorded "live" in a performance venue or in the studio.

As the most ephemeral of all art forms, music happens in the moment, lasting only as long as the composition does. After that, it's gone forever. And every performance is different, even if the musical menu is the same. Only occasionally do all the parts come together in a way that's truly inspired. When that happens, you have musical magic—a transforming experience—which is the essence of recorded music at its best.

Follow any performer on tour and you'll see what I mean. There are the recording engineers at work, night after night, capturing the same sets from start to finish, hoping to catch as many of those magical moments as possible for the CD or music video to follow. If they're lucky, they'll get many over the course of the tour, and the album will really shine.

If you think studio recording is easier, think again. Although artists in this situation don't have the technical sound reproduction problems that can plague large concert halls or for that matter, small venues, they also don't have the give-and-take excitement of the live audience and must generate whatever magic they can…all by themselves.

Given all the trouble that goes into sound recording, it seems important to me that any loudspeaker I listen to should be capable of delivering exactly what was recorded. And I'm doubly insistent when listening to movie soundtracks, which compounds all the dilemmas of musical recording with a wide (and challenging) variety of sound effects. From whispers and pencil tappings…to motorcycles and nuclear explosions...I want to hear it all—the way the filmmaker and recording engineer wanted me to hear it.

That's why I insist on the highest possible degree of loudspeaker accuracy, within the hard limits imposed by the laws of engineering and physics. Furthermore, I think loudspeakers should be able to prove their degree of accuracy with hard measurement data and blind comparisons with live music.

If everybody agreed with me, choosing loudspeakers for your home theatre or surround music system would be a simple matter. You'd be able to do an apples-to-apples comparison of their accuracy characteristics and find the most accurate loudspeaker you could afford. Done.

But not everyone agrees. In fact, loudspeaker accuracy has become a pretty controversial topic in audio circles.

Why? Some people argue that accuracy is in the ears of the beholder. If it sounds good to you, it's good enough, they say.

I disagree. True accuracy is a black and white proposition, not a matter of opinion. It's objective, not subjective. Most math equations, after all, have only one proper solution.

Many people confuse the accuracy issue by using sloppy or incomplete measurements to "certify" their loudspeakers as being accurate.

So, what's a buyer to do? Keep reading. Over the next few months, I'll be using this column to dismantle—bit by bit—the controversy over loudspeaker accuracy. I'll explain the different measurements you should look for when loudspeaker shopping (see sidebar), and help you understand why each
one is important to loudspeaker performance…and to your buying decision.

Designing a truly accurate loudspeaker is a highly complex task, but buying one shouldn't have to be. I'll see what I can do to help. For me—this time, it's personal. WSR

About John Dunlavy

John Dunlavy is a well-known and highly respected engineer and scientist with a number of patents to his credit, and 40 years at the forefront of audio innovation. John's special interest was loudspeaker accuracy, a notion he worked on and talked about for decades. Agree or disagree with him, we can all learn something valuable about the world of audio reproduction, sitting at John's knee.

Part I: Parameters And Elements of Accurate Loudspeakers

Accurate Measurements

The perfect loudspeaker doesn't exist, of course—at least not yet. We're still constrained by the materials available to us and the laws of physics. Still, we have come a long way. This month, I would like to share with you an overview of the performance properties and design features you should look (and listen) for when evaluating the accuracy of loudspeakers. The following 10 topics discuss in brief how to objectively evaluate a loudspeaker for reproducing music and voices. A full set of accurate measurements are essential for determining the potential of a loudspeaker to accurately

reproduce complex musical sounds and transients. Necessary for accurate measurement is an instrumentation-quality microphone used in conjunction with a computer-based measurement program such as MLSSA (Maximum Length Sequence System Analysis), which typically involves the use of 32,768 individual impulses, randomly separated. The computer uses an FFT (Fast Fourier Transform) analysis to obtain all of the measurable properties of a loudspeaker, except for the directivity patterns and impedance. In addition to the use of this program, if maximum accuracy is desired, it is best to also take the measurements within a large anechoic chamber with excellent properties. Why do I put so much emphasis on accurate measurements? It is because there is a correlation between what measurements have to reveal to us versus what we hear. Without measurements, how can you be certain that you've got it when you only think you have it. I'll be discussing each of them in greater depth in the months ahead. I hope the information will be useful to you.

Enclosure Design Properties—External And Internal

Proper enclosure design requires that the designer gives attention to the free-air resonance of the drivers and their "Q," along with the resonance and "Q" of the enclosure. System resonance is a function of the combination of the resonance frequency of the driver and the resonance frequency of the enclosure. Resonance is that frequency of a system at which the reactive component goes from a positive value through zero to a negative value. At the frequency at which the reactive component goes through zero defines the resonance frequency of the drivers and/or the system, i.e., the driver(s) plus its enclosure. It really doesn't matter how esoteric the drivers used in a loudspeaker system are. What does matter is that the drivers are selected for their measurably accurate performance within their frequency response range and their properties are suitably matched to the properties of the enclosure.

At bass frequencies you need a certain volume in the enclosure combined with a driver having an appropriate resonant frequency and "Q" properties. Enclosures that satisfy these properties will yield overall, the most accurate performance. Without either appropriately rounded enclosure edges or effective acoustical absorbent materials between the drivers and the enclosure edges, edge defraction will create significant peaks and valleys in the frequency response of the system, along with ringing in the reproduction of musical transients.

To maintain a symmetrical radiation pattern, both vertically and horizontally, individual drivers or arrays of drivers must be located symmetrically with respect to enclosure edges, otherwise significant peaks and valleys will be created in the horizontal and/or vertical radiation patterns of the loudspeaker.

Impulse, Step, And Phase Response

Impulse Response

An acoustical impulse is a very brief "tic" sound that, surprising to most people, simultaneously contains all audio frequencies. The measured impulse response of a loudspeaker is important because it defines how faithfully a given loudspeaker can reproduce complex or short-duration musical transients without "blurring." While non-technical people may find it difficult to imagine, mathematically, the impulse contains all of the information needed by a computer to accurately derive the frequency response, step response, cumulative spectral decay (waterfall) response, and phase response. Impulse response is usually based upon using a rectangular pulse having a very fast rise time and a width of less than about 20 microseconds.

Step Response

Step response, sometimes referred to as the "transfer-function" of a device, is a measurement that reveals how accurately a loudspeaker can reproduce the shape of a rectangular pulse having a very long duration (greater than 100 milliseconds.) A "perfect loudspeaker" would have no linear or non-linear distortion over the intended frequency spectrum, and would reproduce such a signal with little or no initial overshoot or slope, followed by an exponential roll-off in amplitude versus time, until the amplitude approached zero. Step response reveals a great deal about the overall properties of a loudspeaker because the shape of the reproduced curve provides an easy, visual picture of both amplitude and phase versus frequency, crossover network parameters, and driver time alignment. Therefore, a properly reproduced step response is a reasonable guarantee of a truly accurate sounding loudspeaker, capable of preserving the spectral properties of the original live musical performance, all else being equal.

A well-designed loudspeaker should yield an impulse response with a rapid rise-time in the positive direction, a steep drop from the maximum, followed by an "overshoot" in the negative direction that is more than about 12 to 15 dB below the level of the positive peak. This should be followed by no more than about 200 to 500 microseconds of additional ringing which has a maximum amplitude at least 15 to 20 dB lower than the peak amplitude of the impulse. A large overshoot and/or ringing of substantial amplitude and duration can have the effect of blurring complex musical transients or dulling the impact of fast rise-time musical signals. Ringing can be described as repetition of the original impulse in the time domain. Such pronounced overshoot and/or ringing can also turn a sharp "tic" sound into a dull "toc" sound and significantly alter the quality of plucked strings, orchestral bells, cymbals, castanets, and so forth.

Phase Response

Phase response is related to the time domain response of a loudspeaker, i.e., the ability of the loudspeaker to accurately reproduce complex signals over a wide range of frequencies without any time domain delays or distortion. Phase response is directly related to the impulse and step responses of a loudspeaker and is therefore relevant to the accurate reproduction of complex musical transients. A loudspeaker with excellent frequency and phase response properties should also exhibit good impulse and step responses. Conversely, a loudspeaker with an excellent step response should exhibit a smooth curve of both amplitude and phase versus frequency with no time-domain distortion or irregularities. Thus, a loudspeaker with poor phase response (resulting in a poor transient response) can be expected to create an audible "blurring" of complex musical transients, etc.

Frequency Response

The frequency response of a loudspeaker is the modulus (variations) of amplitude versus frequency. Measured accurately within a good anechoic chamber (preferably in conjunction with a state-of-the-art measurement program such as MLSSA), it reveals a loudspeaker's spectral balance, i.e., its ability or inability to reproduce all frequencies or musical tones with equal intensity or volume. This is an essential property if the reproduced sounds of familiar musical instruments are to be heard as possessing "live" properties.

It is usually considered the most important performance property because it is the easiest for the average person to evaluate when listening to loudspeakers. Anomalies

AT THE INTERSECTION OF DESIGN AND SCIENCE.

Great home theater design isn't just about the equipment. It is about extreme optimization of science of sound reproduction and video technology, to the point where everything disappears, and all that remains is the experience. Intelligible vocals. Powerful but smooth bass. Thunderous special effects. Even response in every seat of the house. And visuals that fill your field of vision, plunging you into the scene. Discover what is possible in our Reference Theater at our Design Center.

14210 NE 20th St.
Suite B, Bellevue,WA 98007
425.440.6206.
Design Center open Monday - Friday 10-5, Saturday and Sunday by appointment.
Visit us online at www.madronadigital.com

HARMAN

in frequency response are most detectable when the direct arrival curve departs from the "average level" by more than about ± 2 dB over a range of more than about one-quarter to one-third of an octave in frequency. (An octave is a 2:1 ratio of frequency, e.g, 100-200 Hz, 1-2 kHz, 4-8 kHz, etc.)

However, truly accurate loudspeakers have amplitude versus frequency curves that fall well within ± 1-2 dB limits, with excursions of less than about ± 1 dB within any given octave (without any amplitude or bandwidth "smoothing"). Some loudspeaker designers will argue that room reflections will modify the frequency response heard by a listener by greater values than this. However, the effect of room reflections, as long as they arrive later than about 5 to 6 milliseconds than the direct arrival sound are generally recognized as such by most experienced listeners.

Cumulative Spectral Decay (Waterfall) Plot

Time domain response is viewed by means of a Cumulative Spectral Decay (waterfall) plot. It provides a sequence of individual frequency response plots, each delayed by a short time interval from the preceding plot. It is another excellent tool for gauging the accuracy of a loudspeaker's ability to reproduce complex musical passages possessing significant transient details without blurring or smearing of complex details in the frequency or time domains.

Directivity

The directivity of a loudspeaker, both horizontally and vertically (at numerous frequencies covering the audio spectrum), can reveal a great deal about a loudspeaker's ability to accurately reproduce complex musical transients within listening rooms containing typical reflective surfaces, e.g., floor, ceiling, and walls. It is graphed with plots at several different angles in both vertical and horizontal planes, usually measured in increments of 5 or 10 degrees, as the loudspeaker is rotated about its central axis. These plots are important for they reveal:

- The size of the "on-axis" angular window (vertically and horizontally) within which accurate reproduction should be optimal. This window is usually defined as the angular limits at which the response at 10 kHz is 3 dB below the response on-axis.
- Symmetry of the main radiation lobe in both horizontal and vertical planes, relative to the listening axis.
- Undesirable high-intensity off-axis radiation "lobes" that might bounce off of a sidewall or ceiling, resulting in an audible deterioration of sound quality and imaging at the listening position.

Harmonic And Intermodulation (IM) Distortion

Harmonic and intermodulation (IM) distortions are created within any network or device, including loudspeakers, that exhibits a non-linear property with respect to the amplitude of a signal. Harmonic distortion consists of even and odd harmonics, which were not present in the original signal. IM distortion is caused by two frequencies interacting with each other to produce sum and difference frequencies, which were likewise not present in the original signal.

Within a loudspeaker driver, most non-linear distortions can be traced to the voice coil not being properly centered in the flux field generated by the magnet and its structure. However, if the voice coil encounters any asymmetry within the magnetic field during the extremes of its physical travel when reproducing loud sound passages, non-linear distortions will be produced.

Crossover Networks

The Crossover Network (often called a "Dividing Network") of a loudspeaker is, in many respects, its "heart." Its purpose is to electrically separate (or divide) the audio spectrum into different frequency ranges, with each range designed to best match the individual frequency requirements of the woofer, bass, mid, and tweeter drivers. For example, if signals encompassing the full audio range were fed to most tweeters, the sound reproduction quality would be terrible and, most likely, the tweeter would not survive without damage for very long.

To accomplish this task, most crossover networks for loudspeakers with multiple drivers covering bass, mid, and tweeter frequencies use several capacitors, inductors, and resistors within relatively complex network configurations to best match the electrical requirements of each driver. A properly designed first-order Butterworth crossover network will permit a loudspeaker with time-aligned drivers (drivers whose voice coils are equidistant from the preferred listening location—typically 10 feet, on-axis) to most accurately reproduce complex musical waveforms with the least amount of waveform or time distortion.

Dynamic Range

The dynamic range of a loudspeaker is the highest sound pressure level (SPL) the loudspeaker can produce "above the noise floor," expressed in dB. The highest SPL that a loudspeaker can reproduce is limited by a given level of non-linear distortion, or before one of the voice coils begins to overheat. The noise floor is set either by the noise of the electronic system, or the noise of the surrounding environment (such as the dishwasher in the other room, the HVAC, the passing truck, etc.)

Limited dynamic range, typified by a compression of amplitude above a relatively low listening level, is usually associated with the production of high levels of non-linear distortion (harmonic and IM). At frequencies above 300 Hz, harmonic and IM distortion levels above about 0.5 percent can usually be heard by most audiophiles. It is usually most easily heard at SPLs greater than about 80 to 90 dB.

Today's advanced loudspeaker design technology provides the capability for designing and manufacturing loudspeakers that can reach short-duration peak SPL levels of 105 to 110 dB without generating audibly bothersome levels of non-linear distortion—dependent, of course, upon the type of music being heard, the "deadness" of the listening room, the properties of the power-amplifier, etc.

Input Impedance

Impedance is the combination of the resistive and reactive components of a load, i.e, loudspeaker load. Impedance versus frequency plots describe whether a loudspeaker represents an easy or a difficult "load" for an amplifier to drive. Many loudspeaker designers pay little attention to the impedance excursions of their loudspeakers, assuming that most of today's solid-state amplifiers exhibit sufficient stability and low internal output impedance to eliminate any interface problems. But many present-day loudspeakers exhibit an impedance curve that swings from as little as 1 ohms to as much as 30 ohms (or greater) over the audio spectrum. Such fluctuations can create serious interface problems with many amplifiers (especially with tube amps with a fairly high internal output impedance), resulting in typical deviations in system frequency response of ± 2 to 6 dB and/or generation of audible non-linear distortion. To provide an "easy impedance load" for most amplifiers, a loudspeaker's impedance should be mostly resistive and should fall within the range of 3 to 10 ohms.

Listening Room Considerations

The acoustical properties of the listening


wh a.
Yep, the impossible just happened.
CES
Consumer Technology Association
Jan. 5-8, 2017 | Las Vegas, NV
Register now at CES.tech
#CES2017
Celebrating 50 YEARS
CES | 1967-2017

room and the location of the loudspeakers within it, relative to the primary listening position, can significantly alter the sound of even the most accurate loudspeakers. Thus, it is important to devote a reasonable amount of time assessing the best location for the loudspeakers and the primary listening chair within the room.

The room should not have too many or too few reflections—neither being good for best listening. Also, symmetry of loudspeaker location within the room and with respect to the primary listening chair can help preserve the original imaging of the recording. Doors and windows should preferably also be in symmetrical locations, relative to the loudspeakers if at all possible. Loudspeaker placement, like in real estate, is all location, location, location. WSR

Part II: How Enclosure Design Affects The Music

Enclosure Properties

The three main components of an audiophile loudspeaker are the enclosure, drivers, and crossover network. For a loudspeaker to exhibit truly accurate measurable and audible performance, all three must be treated in an integral manner, with appropriate levels of design consideration given to each and to how they will perform together as a "system."

All too often, loudspeaker enclosures appear to have been designed by devoting more attention to appearance and artistic merit than to the properties required to achieve a high level of accuracy. This, of course, frequently leads to a loudspeaker that looks great, provides a pleasing sound quality, but falls short of delivering truly accurate measurable and audible performance.

Within this article, I will limit my coverage to sealed enclosures because I believe, if properly designed, they provide the potential to yield the most accurate level of reproduction.

When you port an enclosure, it is no longer able to reproduce bass transients with accurate time domain properties. As a consequence, the transient properties of such a loudspeaker are degraded. Many loudspeaker designers may disagree with this but, as a string bass player for over thirty-some years, I am very sensitive to the reproduction accuracy of bass transients. Ported designs distort the impulse and step response of the woofer, which many designers believe detracts from the accuracy of bass reproduction. This inaccuracy can be seen from the distorted shape visible in a measurement of a loudspeaker's step response.

Among the properties of an enclosure that I believe are most important for achieving measurably and audibly accurate performance are:

1. Proper overall dimensions, internal volume, and correct height-to-width-to-depth ratios to attain the desired low-frequency bandwidth.

2. A vertically symmetrical array of drivers (woofer-midrange-tweeter-midrange-woofer) is required to achieve a symmetrical radiation pattern in the vertical plane and reduce the amplitude level of floor reflections.

3. Precise time-domain (path) alignment of all drivers is required to obtain accurate reproduction of complex musical and other transients (at the normal on-axis listening distance of 10 feet).

4. All drivers should be symmetrically located with respect to enclosure edges along a common vertical axis to achieve a symmetrical radiation pattern in both the vertical and horizontal planes. Otherwise, significant peaks and valleys will be created in the horizontal and/or vertical radiation patterns of the loudspeaker.

5. Use of adequately thick enclosure walls to ensure a high degree of rigidity and reduction of resonances. The enclosure walls should be made of a reasonably dense material of appropriate thickness (minimum 3/4 inch). Medium Density Fiberboard (MDF), while not possessing totally ideal properties, is well established among loudspeaker designers and has earned a good reputation for providing excellent performance at an affordable price.

6. The use of efficient acoustical absorbing material located between the drivers and the external enclosure edges to minimize the effects of "edge diffraction" (re-radiation from enclosure edges). Merely "rounding the edges" of an enclosure usually does little to reduce the audible and measurable effects of diffraction or reflections from enclosure edges and protruding driver flanges/edges. Some loudspeaker designers have succeeded in mitigating the diffraction problem by locating the midrange and tweeter drivers on the front baffle surface in an asymmetrical manner so that the distance from the center of a driver to each of the enclosure edges is not divisible into any of the other distances by a "whole number" (integer). While this "solution" may result in a reasonably smooth curve of on-axis frequency response, the off-axis frequency response is usually riddled with peaks and valleys, and the directivity patterns become asymmetrical (with lobes at undesirable angles), often no more than a few degrees off-axis.

7. Separate, individually-sealed, rigidly-constructed, internal enclosures for the woofer, bass, midrange, and tweeter drivers is necessary to prevent physical/acoustical interaction between drivers operating over different frequency ranges.

8. Sufficient internal volume within the woofer and/or bass sections to achieve the intended low-frequency performance. At bass frequencies, a certain volume in the enclosure combined with a driver having an appropriate resonant frequency and "Q" properties is required to yield overall the most accurate performance.

Proper enclosure design requires that the designer gives attention to the free-air resonance of the drivers and their "Q," along with the resonance and "Q" of the enclosure. System resonance is a function of the combination of the resonance frequency of the driver and the resonance frequency of the enclosure. Resonance is that frequency of a system at which the reactive component goes from a positive value through zero to a negative value. At the frequency at which the reactive component goes through zero defines the resonance frequency of the drivers and/or the system, i.e., the driver(s) plus its enclosure. It really doesn't matter how esoteric the drivers used in a loudspeaker system are. What does matter is that the drivers are selected for their measurably accurate performance within their frequency response range and their properties are suitably matched to the properties of the enclosure.

9. Sufficient internal bracing and stiffness to prevent "modes" (maximum and minimum sound pressure levels) from developing along enclosure walls.

10. Use of efficient acoustical absorbing material within the interior chambers of the enclosure housing the woofer, bass, and midrange drivers, in a quantity adequate for achieving proper "damping" of internal standing waves.

Without either appropriately rounded enclosure edges or effective acoustical absorbent materials between the drivers and the enclosure edge, edge defraction will create significant peaks and valleys in the frequency response of the system, along with "ringing" (amplitude variations in the time domain) in the reproduction of musical transients.

While "ported" enclosures are frequently used by some designers to "enhance" the SPL of woofer or bass drivers, it raises the "system Q"—resulting in bass that is often "poorly damped." Indeed, ported enclosures frequently yield what many listeners refer to as "one note bass." (Although such sound quality appeals to many listeners of music that contains mostly "kick drum" bass, it is hardly appropriate for listening to a wide variety of music or soundtracks containing various kinds of bass instruments or sound effects containing complex bass transients.)

Indeed, there are very few (if any) "free lunches" available to designers, as is evident from what competent physics and engineering teach about the properties of loudspeaker enclosures.

The acoustical shortcomings of a poorly-designed enclosure are usually evident from an informed examination of a full set of accurate anechoic chamber measurements that include impulse response, step response, and cumulative spectral decay (waterfall) response. For example, the impulse response of a poorly-designed enclosure will often exhibit a "ringing" characteristic, while the step response will disclose a distorted "leading edge" and undulations along the length of the step. These deficiencies are typically audible to those familiar with the sound of live instruments such as the string bass, cello, drums, etc.

This is why a full set of accurate anechoic chamber measurements (not outdoors and made at the normal listening distance of 10 feet) are so important to audiophiles and others seriously concerned about the true accuracy of their loudspeakers. These should include frequency response, impulse response, step response, waterfall, and non-linear distortion at various frequencies and SPLs; input impedance; and representative radiation patterns.

In future articles, I will discuss the wide range of different loudspeaker measurements and how each can help audiophiles determine whether a given loudspeaker possesses the potential to yield "true audible accuracy." **WSR**

Part III: How Impulse, Step And Phase Response, Frequency Response, Waterfall Measurements, And Input Impedance Affect The Music

The Necessity Of An Accurate Set Of Anechoic Chamber Measurements

It is probably difficult for the average audiophile, perhaps lacking an appropriate background in mathematics, to understand that from a measurement of the "impulse-response" of a loudspeaker, a computer can calculate its frequency response, step response, phase response, etc. This is accomplished using a FFT (Fast Fourier Transform) program, such as MLSSA (Maximum Length Sequence Speaker Analysis) designed by Doug Rife of DRA Labs. It is not an inexpensive program but, if used regularly and properly, is worth every cent it costs—and then some.

However, many of the same measurements can also be made within a good, large anechoic chamber (like the two chambers I built at Dunlavy Audio Labs—DAL) that are approximately 30 feet long, 20 feet wide, and 16 feet high, with all internal surfaces covered with large wedges of high-density open-cell acoustical foam.

But, even with two large anechoic chambers, MLSSA is still used to ensure the highest possible accuracy for all DAL loudspeaker measurements.

Some loudspeaker designers attempt to take their measurements outdoors, but it is seldom that the "ever-present wind," even when it is difficult to "feel," drops to a low enough velocity not to affect the accuracy of measurements. Many loudspeaker designers (and magazines) attempt to circumvent this problem by measuring the bass response of their loudspeakers indoors at a distance of only a few feet (typically placing the instrumentation microphone almost up against the woofer cone) and "splicing" the data to measurements made at higher frequencies over a greater distance (typically one meter). But realistically, nobody listens to a large-size loudspeaker at such a close distance, and therefore such a near-field measurement will typically not convey the loudspeaker's performance at the normal listening distance of 8 to 12 feet. [And, for several other good technical reasons, this seldom yields a set of measurements with reliable and/or consistently usable accuracy.]

Impulse, Step, And Phase Response

An acoustical impulse is a very brief "tic" sound that, surprising to most people, simultaneously contains all audio frequencies. The measured impulse response of a loudspeaker is important because it defines how faithfully a given loudspeaker can reproduce complex or short-duration musical transients without "blurring." While non-technical people may find it difficult to imagine, "mathematically," an impulse contains all of the information needed by a computer running a FFT program to accurately derive the frequency response, step response, cumulative spectral decay (waterfall) response, and phase response. Impulse response is usually based upon using a rectangular pulse having a very fast rise time and a width of less than about 20 microseconds. (See Figure 1, Impulse Response)

A well-designed loudspeaker should possess an impulse response with a rapid rise-time, a steep drop from the maximum, followed by an "overshoot" in the negative direction that is more than about 12 to 15 dB below the level of the positive peak. This should be followed by no more than about 200 to 500 microseconds of additional ringing (with a maximum amplitude at least 15 to 20 dB lower than the peak amplitude of the impulse). Ringing is an undesirable characteristic for all audiophile components including loudspeakers, amplifiers, pre-

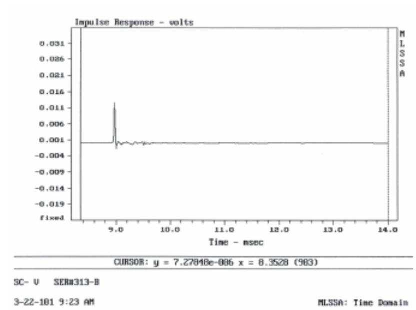

Figure 1 Dunlavy Signature Series SC-V

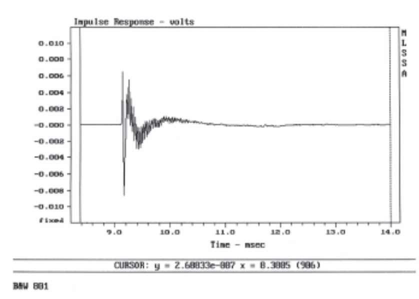

Figure 1 An Expensive Audiophile Loudspeaker

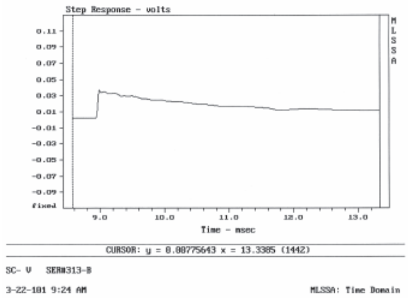

Figure 2 Dunlavy Signature Series SC-V

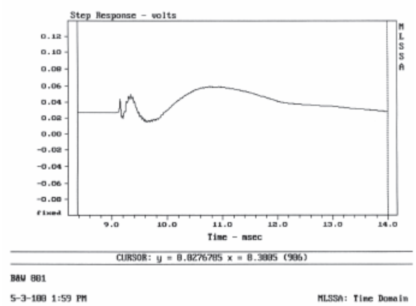

Figure 2 An Expensive Audiophile Loudspeaker

amps, signal sources, etc. A large overshoot and/or ringing of substantial amplitude and duration can have the effect of blurring complex musical transients or dulling the impact of fast rise-time musical signals. Ringing can be described as repetition of the original impulse in the time domain. Such pronounced overshoot and/or ringing can also turn a sharp "tic" sound into a dull "toc" sound and significantly alter the quality of plucked strings, orchestral bells, cymbals, castanets, and so forth.

Step response, sometimes referred to as the "transfer-function" of a device, is a measurement that reveals how accurately a loudspeaker can reproduce the shape of a rectangular pulse having a very long duration (greater than 100 milliseconds). A "perfect loudspeaker" would reproduce such a signal with little or no initial overshoot, followed by an exponential roll-off in amplitude versus time, until the amplitude eventually approached zero. (See Figure 2, Step Response)

Step response also reveals a great deal about the overall properties of a loudspeaker because the shape of the reproduced curve provides an easy, visual picture of both amplitude and phase versus frequency, crossover network parameters, and driver time alignment. An accurate step response reveals whether a loudspeaker is truly accurate with respect to its frequency response, phase response, impulse response, and waterfall response. Therefore, a properly reproduced step response is a reasonable guarantee of a truly accurate sounding loudspeaker, capable of preserving the spectral properties of the original live musical performance, all else being equal.

Phase response is related to the time domain response of a loudspeaker, i.e., the ability of the loudspeaker to accurately reproduce complex signals over a wide range of frequencies without any time domain delays or distortion (including the ability of a loudspeaker to accurately reproduce square waves). Phase response is directly related to the impulse and step responses of a loudspeaker and is therefore relevant to the accurate reproduction of complex musical transients. A loudspeaker with excellent frequency and phase response properties should also exhibit good impulse and step responses. Conversely, a loudspeaker with an excellent step response should exhibit a smooth curve of both amplitude and phase versus frequency, with no time domain distortion or irregularities. Thus, a loudspeaker with poor phase response (resulting in a poor transient response) can be expected to create an audible "blurring" of complex musical transients, etc.

Frequency Response

The frequency response of a loudspeaker is the modulus (variations) of amplitude versus frequency. Measured accurately within a good anechoic chamber (preferably in conjunction with a state-of-the-art FFT measurement program such as MLSSA), it reveals a loudspeaker's spectral balance, i.e., its ability or inability to reproduce all frequencies or musical tones with equal intensity or volume. This is an essential property if the reproduced sounds of familiar musical instruments are to be heard as possessing "live" properties.

It is usually considered the most important performance property because it is the easiest for the average person to evaluate when listening to loudspeakers. Anomalies in frequency response are most detectable when the direct arrival "curve" departs from the "average level" by more than about ± 2 dB over a range of more than about one-fourth to one-third of an octave in frequency. (An octave is a two-to-one ratio of frequency, e.g, 100 to 200 Hz, 1 to 2 kHz, 4 to 8 kHz, etc.) (See Figure 3, Transfer Function Magniture—Frequency Response)

However, a truly accurate loudspeaker should possess an amplitude versus frequency curve that falls well within ± 1 or 2 dB limits, with excursions of less than about ± 1 dB within any given octave (without any amplitude or bandwidth "smoothing.").

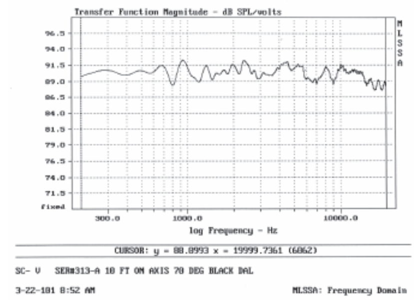

Figure 3 Dunlavy Signature Series SC-V

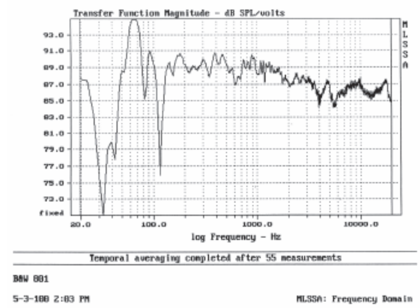

Figure 3 An Expensive Audiophile Loudspeaker

MLSSA provides a "smoothing function" that allows the measurer to manipulate a frequency response that measures ± 3 to 5 dB and make it look like one that appears to be only ± 1 or 2 dB. Some designers will argue that room reflections will modify the frequency response heard by a listener by greater values than this amount. However, the effect of room reflections, as long as they arrive more than about 5 to 10 milliseconds (sound travels approximately one foot per millisecond) later than the direct arrival sound, are generally aurally processed and recognized as such by most experienced listeners.

Waterfall Measurements

Time domain response is viewed by means of a Cumulative Spectral Decay plot (often called a waterfall plot). It provides a sequence of individual frequency response curves, each delayed by a short time interval from the preceding plot. It is another excellent tool for gauging the accuracy of a loudspeaker's ability to reproduce complex musical passages possessing significant transient details without blurring or smearing of complex details in the frequency or time domains. (See Figure 4, Cumulative Spectral Decay—Log Frequency—Hz)

Input Impedance Versus Frequency

The ability of a loudspeaker to be "amplifier-friendly" is often related to the loud-

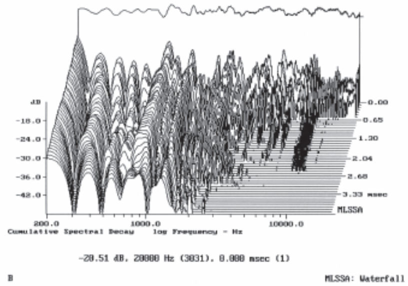

Figure 4 Dunlavy Signature Series SC-V

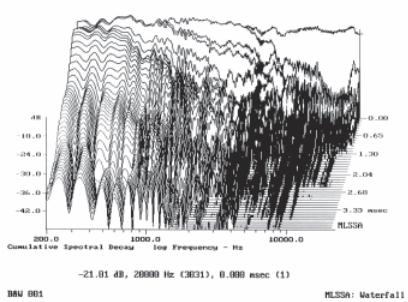

Figure 4 An Expensive Audiophile Loudspeaker

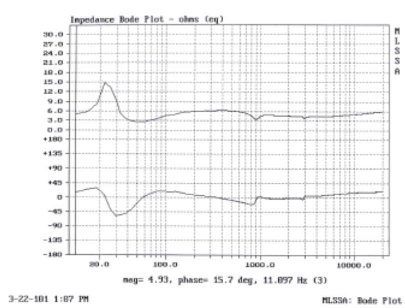

Figure 5 Dunlavy Signature Series SC-V

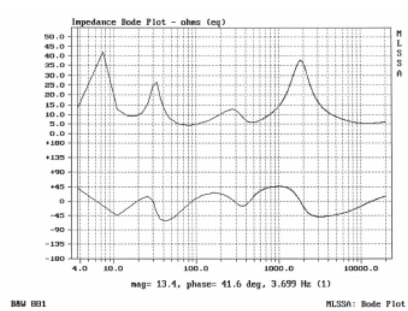

Figure 5 An Expensive Audiophile Loudspeaker

speaker's input impedance versus frequency. This is especially true with regard to tube amplifiers and some solid-state amplifiers using minimal inverse-feedback. This is because the internal output impedance of such amplifiers is often higher than 1 ohm, which when feeding a loudspeaker whose input impedance might vary from 1 or 2 ohms to a maximum of 20 or 30 ohms or more, can result in such a system no longer possessing a flat frequency response (amplitude versus frequency and/or accurate impulse and step responses). A well-designed power amplifier should have an output impedance lower than a few tenths of 1 ohm because a higher output impedance, feeding a loudspeaker with an input impedance varying from 2 ohms to perhaps 30 ohms (or more) can potentially create audible fluctuations in frequency response. Therefore, a well-designed loudspeaker should have an input impedance that does not drop below about 3 ohms over the audio spectrum, especially at higher frequencies, e.g., 3 to 5 kHz. It is necessary, for accurately evaluating a loudspeaker's input impedance with respect to its being amplifier-friendly, to have available both its resistive and reactive components. This is because many tube amplifiers, or other Class A amplifiers, that exhibit a relatively high output impedance can alter the frequency response and time domain response of a system using loudspeakers having an input impedance that varies significantly with frequency. (See Figure 5, Impedance Bode Plot—Ohms).

Conclusion

While loudspeakers that do not measure accurately may sound musically good, satisfying, etc., they can never reproduce music with the audible accuracy of loudspeakers that measure truly accurate in all of the above mentioned categories. WSR

Part IV: How Directivity, Distortion, Crossover Design, Input Impedance, Dynamic Range, And Imaging

Directivity

A lot of controversy exists regarding the optimum directivity properties for loudspeakers.

Some designers claim that loudspeakers with a near omni-directional pattern provide the most accurate reproduction. But few musical instruments exhibit omni-directional radiation characteristics, especially at higher audio frequencies, e.g. above about 1 kHz (where a wavelength is about 1 foot).

Thus, within listening rooms possessing average acoustical properties, it would appear that the "audibly most accurate reproduction" would most likely be achieved with loudspeakers whose radiation patterns most closely emulate an "average" of those exhibited by most musical instruments, as a function of frequency. By achieving this, a loudspeaker's reproduction (if properly located within the listening room and equidistant from the listening position), should most closely approximate the overall sound properties of the original sound.

Carefully controlled, "blind" comparisons of loudspeakers with several musical instruments within typical "audiophile" listening rooms, has consistently proven this to be true.

The directivity of a loudspeaker, both horizontally and vertically (at numerous frequencies covering the audio spectrum), can reveal a great deal about a loudspeaker's ability to reproduce complex musical transients within listening rooms containing typical reflective surfaces, e.g., floor, ceiling, walls, furniture with hard surfaces, etc.

Directivity patterns also reveal a great deal about a loudspeaker's ability to provide accurate imaging that does not cause notes at different frequencies from the same musical instrument to appear to arrive from a different direction. It is typically graphed with plots of "amplitude versus frequency" at several different angles in both vertical and horizontal planes, usually measured in increments of 5 or 10 degrees, as the loudspeaker is rotated about its central axis. These plots are important for they reveal:

1. The size of the "on-axis" angular window (vertically and horizontally) within which the loudspeaker's reproduction is the most accurate. This window is often defined as the angular limits at which the one-third-octave response at 10 kHz is 3 dB below the response on axis.

2. Symmetry of the main radiation lobe in both horizontal and vertical planes, relative to the listening axis. (Asymmetrical radiation patterns often result in inaccurate stereo imaging.)

3. Undesirable, high-intensity, off-axis radiation "lobes" (side lobes) that might bounce off of a sidewall, ceiling, etc., resulting in both an audible deterioration of