Audio *perfectionist* Journal

Issue #5

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This issue builds on concepts that were presented in previous Issues. It's not absolutely necessary to have read the first four issues in order to benefit from the information offered in this one, but it's a good idea.

Audio perfectionist Journal

Issue #5

This is Issue #5 of the Audio Perfectionist Journal. In this issue we will begin to examine loudspeakers. We'll discuss speaker accuracy because high fidelity reproduction is our stated goal.

Speaker accuracy can be measured by objective tests. These tests can't tell you everything about how a particular speaker model will sound but they can tell you if a certain design has the potential for accurate reproduction.

Speakers are very important components in any audio system. Many people believe that speakers are so important that they spend too much on them and are then forced to compromise on the quality of the components that create the signal which is fed to the speakers. These folks hope that their carefully chosen speakers will somehow improve that signal and produce good sound. This is, of course, impossible. Speakers can't improve the quality of a bad signal but that doesn't mean that you can put a perfect signal into an inaccurate speaker and get good sound either. In fact, inaccurate speakers will produce inaccurate sound regardless of the quality of the signal that is fed to them.

Poorly designed speakers can fool all of the people some of the time and some of the people all of the time, but they can't fool all of the people all of the time. Examining speaker test graphs won't allow you to choose a satisfying speaker without listening, but those graphs may help you avoid a costly mistake. Why risk being momentarily fooled by a speaker which can't possibly create an accurate replica of the input signal? This issue of the Audio Perfectionist Journal (and the next one) will help you separate the posers from the contenders.

In This Issue

The article titled *The Truth About Loudspeakers* describes what an accurate speaker is and what an accurate speaker should do. This article discusses why most speakers are demonstrably inaccurate and the reasons why most fail to provide output response that is an acoustic replica of the electrical input signal.

The article titled *Meaningful Speaker Measurements* describes the various speaker tests which I believe are useful to consumers who are evaluating speakers for purchase. Most of us don't want to design speakers; we simply want to get the highest performance possible and the best value for our money. Measurements can tell us which products are worthy of our consideration and which are best left to those who listen only to background music at elevator levels.

The article titled *Interpreting Speaker Response Tests* explains how to interpret the graphs that various magazines use to dress up their reviews. These graphs are designed to add a sense of scientific credibility to reviews written by people who seldom have any real technical knowledge but, if you learn how to interpret the test graphs, you may actually benefit from the data they contain.

Most magazines don't want you to know too much. If they would fully explain what their test graphs mean they could risk offending big advertisers whose poorly designed speakers are often exposed by the measurements that accompany the rave reviews they publish. After you read this issue you'll be able to interpret the measurements published by the magazines and come to your own conclusions.

Coming Up

This issue is all about speakers. The popular speakers which are measured and discussed in this issue are demonstrably inaccurate in one way or another. In the next issue we'll continue to examine speakers, but the models measured and discussed in issue #6 are demonstrably accurate within all the currently accepted "objective" parameters. After you see that many available speaker models can accurately reproduce the input signal, we'll talk about sound. Yes, even those speakers which are truly accurate don't all sound alike.

You can eliminate the poorly designed products, and those products which are skillfully designed to be inaccurate on purpose, by examining objective test results. Tests won't define all the audible differences between those speaker models which are competently designed to be accurate electromechanical transducers. Even skillful engineering requires a balance of compromise. In issue #6 we'll compare some accurate speakers and examine the various compromises chosen by their designers. I'll tell you what I hear when I compare these speakers but you'll have to listen for yourself to determine whether any of these models will satisfy your personal tastes.

In this issue you'll see that spending a lot of money doesn't guarantee that you'll get an accurate, well designed product. In the next issue you'll discover that truly accurate speakers can be purchased for less than one thousand dollars a pair and that some of the finest speakers available can be owned by regular people like you and me without taking out a second mortgage.

All the information in issues #5 & #6 is specifically about the speakers used as a stereo pair or for the front left and right channels in a home theater system. In future issues we'll discuss speakers for home theater center channel and surround use.

The Truth About Loudspeakers

The truth about loudspeakers is that the vast majority of commercially available designs can't come close to accurately reproducing the electrical signal created by your expensive audio components. Most speakers were designed to be inaccurate on purpose in order to create a unique sound. Many engineers who do attempt to design accurate speakers ignore time, phase and energy storage as if these factors have no audible effects—they strive for flat frequency response above all else. Some speakers are purposely designed to increase time smear by directing a large portion of their output towards the room boundaries. Some speaker types store and release so much energy that they add artificial reverberation to the sound. All this results in a marketplace filled with a huge variety of speaker types, most with a unique sonic signature and almost none with the ability to accurately convert the electrical input signal to acoustic energy—the true goal of an accurate transducer.

An accurate loudspeaker should not have a sonic signature or a distinct sound. An accurate speaker should convert electrical energy to mechanical energy without adding or subtracting anything. The acoustical waveform coming out of the speaker should be an exact replica of the electrical waveform entering the speaker at the input terminals. Very few commercially available speaker systems can accomplish this difficult task, and this article will discuss why this is true and how you can tell which speaker models are worthy of consideration if sonic accuracy is what you seek.

Let's examine some of the factors that make one speaker model sound different from another and then we'll discuss some measurements which objectively gauge some of these differences.

Frequency Response

Flat frequency response is an absolute requisite for accurate musical reproduction. All tones must be reproduced at the correct volume and the harmonic balance of each note must be preserved. Incompetently designed loudspeakers, which have large deviations from flat frequency response, cannot accurately reproduce music. Some speakers from competent designers can't either. Some competent engineers design speakers with large deviations from flat frequency response on purpose.

In an industry crowded with manufacturers making more products than consumers will buy, many companies have resorted to creating speakers with a distinctive "sound" in order to set their products apart from all the rest. The simplest way to do this is to tailor the frequency response of the loudspeaker in order to spotlight certain portions of the frequency spectrum. This spotlighting may appear to the uninitiated as an improvement in performance over a certain range of frequencies. What range? Well, that depends on who the designer is trying to seduce.

Car stereo systems emphasize mid bass, often at just one frequency centered around 80Hz. There are some extremely expensive "high-end" audio speakers that do basically the same thing. (See my first letter to Stereophile.) Speakers which are designed to fool magazine reviewers and neophyte audiophiles may actually de-emphasize bass and emphasize various frequencies in the midrange and highs. Spotlighting upper frequencies may convince an inexperienced listener that he is hearing more "detail" or "air."

A loudspeaker system that emphasizes some frequencies and de-emphasizes others may sound spectacular when reproducing certain recordings but, based on my experience, it will not provide long term satisfaction to those of us who listen to a variety of music. If you use inaccurate speakers, you limit your choice of listening material to those recordings which are "enhanced" by the tailored response of the speakers. Rap "music" may be flattered by car stereo response but a solo cello performance will not be.

No speaker can be considered to be an accurate transducer unless it has flat frequency response within narrow limits. In my opinion, these limits should be less than ±3dB. With frequency response deviations greater than ±3dB, a loudspeaker will emphasize some notes or harmonics, and de-emphasize others. Even if the useful bandwidth of the speaker is limited, the response within this bandwidth should be flat. The Quad ESL speakers that satisfied my musical lust for so many years had little output at either end of the frequency scale. That didn't prevent them from delivering an immensely satisfying musical experience because the bandwidth that they could reproduce was accurately rendered in both frequency and phase.

M&K speakers have flat frequency response within ±2dB according to their literature and Aerial speakers do, too. Why do M&Ks and Aerials sound so different? Does a .5dB peak here and a 1dB dip there make that much difference in overall sound? The answer is yes and no.

You can make substantial changes in the sound of a loudspeaker by applying equalization in very small amounts. Changes as small as .5dB (one half dB) in the midrange can be easily heard by most experienced listeners. Frequency response is very important to be sure but other factors may be even more significant. Most of these other factors are temporal in nature and most have been completely ignored by a scientific community that has been obsessed by the idea that frequency response is <u>all</u> that matters. Frequency response is important but so are other factors. Consider music for a moment.

Music, Intensity and Time

While flat frequency response is an absolute requisite for accurate musical reproduction, it's not the only thing that matters. Music is more than melody and non-linear frequency response is only one of the factors that can screw up musical reproduction. Ivor Tiefenbrun, the founder of Linn Products in Scotland, has stated that there are only two things that separate music from noise: the tune and the beat. Music has a melody of course, but virtually all other aspects of music are temporal in nature.

Rhythm is obviously temporal. To create music you have to play the right note at the right time with the correct pace. If all the notes are played correctly but not at the correct times, the song will be drastically altered. The results may still sound musical to some but certainly won't represent the intentions of the composer. Altering musical pace can turn a song of celebration into a funeral dirge.

Every musical note is composed of a fundamental tone accompanied by harmonic overtones. A piano and a violin can play the same note with the same fundamental frequency and still sound completely different because the harmonic structure of the sound that each instrument produces is completely different. Altering the amplitude ratios between the fundamentals and harmonics, as a speaker with inaccurate frequency response would, alters the perceived pitch and the tonal character of the sound. Altering the temporal relationships between the fundamentals and the harmonics, as a speaker that scrambles phase and time relationships would, alters the tonal character of the sound, too.

Intensity over time is critical. A musician plays some notes louder and some notes softer as a part of the interpretation of the composer's intent. The characteristic sound of an instrument is related to the time it takes to reach peak intensity and how long that intensity is sustained. The perception of intensity is affected by time. How quickly the peak amplitude is achieved is as important as the amplitude of the peak. The intensity of the harmonic content affects the timbre of the note and the perceived pitch. How quickly the peak ebbs—reduces in intensity over time—is another important characteristic of the unique sound of each instrument.

Many different instruments in the orchestra can play the same notes at the same frequencies. They produce different sounds due to differences in harmonic structure, differences in intensity over time and the way in which the sound vibrations are initiated. Accurate speakers must reproduce all the elements that differentiate instrumental sounds without adding or subtracting anything. This is a tall order.

Most experienced listeners agree on the importance of accurate frequency response but time domain performance is just as important, in my opinion. I can tell you from experience that speakers which scramble the phase of fundamentals and harmonics (as all speakers with steep-slope crossovers do) change the tonal character and perceived pitch of music and scrambling phase also has a negative impact on image focus. Once you become accustomed to time- and phase-accurate speakers, you can never go back.

It seems obvious to me that the ear/brain mechanism is at least as sensitive to time and intensity as it is to frequency but this contention is certainly not universally accepted and is a matter for debate. I'll make a case here for my position but readers are advised to consult the literature—and there is a ton of it—for opposing viewpoints. You should draw your own conclusions based on your own observations because you'll experience reproduced sound in the same unique way that you experience real sound.

Transfer Function

The transfer function of a component describes how the signal is altered when passing through this component. A transfer function test compares the amplitude (frequency response) and phase of the output signal to the amplitude and phase of the input signal to see what effects the component under test had on the amplitude *and phase* of the signal that passed through it. Note that the transfer function considers phase as well as amplitude-with-frequency to determine what the component under test did to the signal besides what it was supposed to do. As an example, an amplifier should make the signal bigger without changing the frequency or phase characteristics of the signal at all. Commensurately, an accurate speaker should create an acoustical replica of the electrical input signal without changing the frequency or phase characteristics of the signal at all. While this concept is incredibly simple and logical, most speaker designers ignore it.

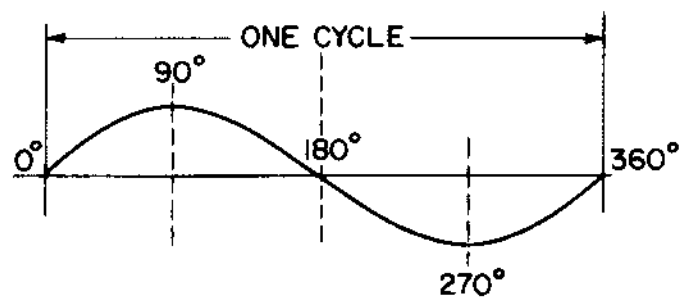
A transfer function graph displays how well the output signal tracks the input signal in both amplitude-with-frequency and phase. The transfer function of a loudspeaker will tell you a lot about what that speaker sounds like. Speakers with similar transfer functions *do* sound similar, but there's more, as we'll see.

Transfer function considers frequency response and phase response. These two elements reveal a lot about a loudspeaker's performance capabilities in reproducing intensity over time—the most important aspects of accurate musical reproduction in my opinion. In the speaker response graphs that we will be examining, frequency response and phase response will not be shown together on the same plot. We'll interpret the time- and phase-accuracy of the speakers tested from the impulse and step response graphs.

We all know that a frequency response graph shows us the amplitude of the output signal of the device under test, across its range of frequencies. The input signal for the test has a constant amplitude regardless of frequency. The output signal, a measurement of which is used to create the graph, shows if the device under test emphasizes or de-emphasizes any frequencies within its useful bandwidth. A straight line across the graph would indicate accurate response, and deviations from a straight line indicate errors in response. If the graph shows that the signal is 2dB higher in amplitude at 4kHz than it is at 1kHz, the device under test emphasizes signals at 4kHz. If the graph shows that signal amplitude is 3dB lower at 2kHz than it is at 1kHz, the device under test de-emphasizes signals at 2kHz. This is pretty simple and easy to understand. Phase is a little more difficult to interpret. So, just what is phase and how is it relevant?

Phase

Phase is a mathematical concept that describes the timing of an electrical waveform. Points on an alternating current waveform can be described in degrees like points on the circumference of a circle. A complete cycle of the wave includes 360 degrees. The position (or time) of various points on the wave during a single cycle can be described as being in a range of from 0 to 360 degrees (or smaller increments) from the beginning of the cycle. If you know the frequency of the wave, then you know the period of the cycle. If you know the period of the cycle you can determine the time difference that a phase change will create.



Two electrical waves are said to be "in phase" when the peaks and valleys of each wave coincide. Electrical waves at the same frequency can be out of phase by varying amounts and the phase lead or lag can be described in degrees (or smaller increments) and the time lead or lag can be calculated. Electrical waves at the same frequency can be in phase yet out of time if they differ in phase by increments of 360 degrees. In other words, two waves can be in phase yet out of time by one complete period.

Acoustical phase has to do with acoustical pressure waves rather than electrical waves. These pressure waves are more usually described in terms of distance rather than degrees. If you know the frequency of a sound pressure wave you can calculate the "wavelength" of each cycle. A complete cycle rises to a pressure peak, falls to a pressure minimum and returns to the nominal room pressure where it began. When the air pressure wave reaches your ears it moves your ear drums. As the air pressure increases your ear drums are momentarily moved inward and when the pressure decreases your ear drums are momentarily moved outward. We perceive these changes in air pressure as sound if they occur at frequencies within our range of hearing.

Two five-inch sound waves are in phase if the pressure peaks coincide at the microphone or at the listener's ear. They are out of phase if the pressure peak of one wave is coincident with the pressure minimum of the other wave at the microphone or the listener's ear. Out-of-phase sound waves may cancel each other. If the pressure peak of one wave equals the pressure drop of the other, your ear drums will not move and you will perceive no sound.

Phase is a way of describing when something happens in relation to something else. It is an expression of relative time that relates to an electrical or acoustical waveform. What elements in a loudspeaker design have the greatest effect on phase? The crossover network that divides the electrical signal into frequency ranges has the greatest effect on electrical phase and the physical position of the various drive elements in relation to the listener has the greatest effect on acoustical phase.

Both acoustical phase and electrical phase affect when various parts of the signal produced by a loudspeaker arrive at the listener's ears. A speaker that is time- and phase-correct can deliver all frequencies to the listener at the same time. Speakers with steep-slope crossover networks cannot do this, even if the drivers are physically offset as in the B&W 801. With a steep-slope crossover network, signals reproduced by the tweeter will arrive first, followed by signals reproduced by the midrange driver, followed by signals reproduced by the woofer. Speakers with all drivers mounted on a flat, vertical baffle can't deliver all frequencies to the listener at the same time either—even if they have first-order crossovers like some Dynaudio models. Because the acoustic centers of the drivers are not aligned, the sounds produced by each driver will arrive at the listener's ears at different times.

Crossover Networks

Speakers with multiple drive elements—woofers, midranges and tweeters—have electrical networks to divide the frequency range of the input signal into segments. The crossover network divides up the full range electrical signal and allocates portions to the specific driver assigned to reproduce that range of frequencies. Low frequencies go to the woofer, midrange frequencies go to the midrange driver, and high frequencies go to the tweeter. The crossover network adjusts the output levels of the drivers so that each frequency range is reproduced at equal volume. Crossover networks also may equalize response nonlinearities in the drive elements and compensate for driver impedance variations.

Crossover networks create phase shift. The steeper the slope, the greater the shift. The capacitors and inductors used in crossovers create phase shift in opposite directions. Series capacitors are used in high-pass filters and series inductors are used in low-pass filters. Phase is gradually altered as reactive components become reactive—it changes with frequency. Voltage leads current in an inductive circuit and current leads voltage in a capacitive circuit. This gets intensely complicated so I'm going to use some generalizations here to give you an idea of what goes on.

In general, each order of filter slope causes about 90 degrees of phase shift. This variable delay changes with frequency and, when filter slopes go beyond first-order (more than 6dB/octave) the delay can't be fully compensated for by repositioning the drive elements, as we'll see when we start to examine step response graphs.

Filters store and release energy. Filters with multiple reactive elements are resonant in nature. As filter slopes get steeper, transient response is degraded because steeper filters ring (oscillate) more after the signal stops. The steeper the slope the greater the ring, as we'll see when we start to examine impulse response graphs.

A first-order electrical crossover network can have as few as two parts: a single capacitor for high-pass and a single inductor for low-pass. A first-order speaker will be far more complicated. First-order speakers have acoustical response that is tailored to produce a first-order transition between drive elements. The crossover network may need to be very complex in order to accomplish this 6dB/octave acoustical roll-off because the electrical and mechanical characteristics of the drivers must be considered.

Timing

Each individual drive element in a loudspeaker reproduces a portion of the frequency spectrum. Different drivers may be called upon to reproduce parts of a musical waveform. The waveform may represent a single note played by a single instrument. In certain instances, reproducing a single note may require that the woofer reproduce the fundamental frequency while the midrange and tweeter reproduce portions of the harmonics at higher frequencies. For accurate reproduction it is important that the amplitude relationships between the fundamentals and harmonics not be altered, as they would be by a speaker with inaccurate frequency response. It is just as important that the time relationships between the fundamentals and the harmonics not be altered, as they would be by a speaker that scrambles time and/or phase.

To maintain the integrity of the waveform the output of all the drive elements must be timed to arrive at the listener's ears at exactly the same moment. This is more difficult than it first appears to be. Woofer drivers are large and have relatively massive moving parts. Tweeters are small and the moving parts are very light. Midrange drive elements fall somewhere in between. Tweeters must have a very short "rise time" in order to reproduce high frequencies which, by definition, require faster response. Woofers reproduce low frequencies where rise time seems to be of less concern but some woofers are still faster than others.

No woofer is as fast as the slowest tweeter. The time that elapses between the arrival of the electrical signal at the input of the driver and when the acoustical signal leaves the driver diaphragm will vary depending on the size and frequency range of the driver. Tweeters are faster than midrange drivers which are faster than woofers. This is only a small part of the story, however.

Drivers operate on electromagnetic force. The strength of an electromagnetic force is measured in ampere turns. Electromagnetic force is created by current and inductors resist a change in current. Voice coils are iron-core inductors and larger drivers generally have more voice coil inductance. Series inductors are used to make low-pass filters. Woofers generally have low-pass filters and so do midrange drivers.

Tweeters have small diaphragms which are usually shaped like a convex dome. Woofers have large cone-shaped diaphragms with the apex of the cone facing inward. The acoustic center of a tweeter will usually be in line with, or forward of, the flange which is used to attach the driver to the front baffle of the speaker. The acoustic center of a woofer will usually be somewhat behind its mounting flange (maybe several inches behind). Midrange drivers fall somewhere in between.

All these factors add up to produce a delay in the output from various drivers relative to the input signal. In general, the lower the frequency range of the driver and the larger the driver, the greater the delay. In a first-order speaker these delays can be easily compensated for by stepping the midrange driver back from the woofer to compensate for the greater delay from the woofer, and stepping the tweeter back from the midrange to compensate for the greater delay from the midrange. No amount of physical offset can fully compensate for the phase shift caused by steep-slope crossovers because the drivers appear to change physical position with frequency due to the effects of the crossover networks. First-order speakers can be fully time-compensated by offsetting the drive elements, as we'll see when we examine step response graphs of time- and phase-accurate speakers in the next issue. Speakers with steep-slope crossovers cannot be fully time-compensated by physically offsetting the drivers.

Stored Energy

Music is transient in nature. Notes may follow in quick succession or there may be periods of silence where the decay of the instruments and the ambient sound of the environment become important elements of the performance. Loudspeakers should not sing along after the tune has ended. If they do, many subtle elements of the music will be obscured. Musical instruments store and release energy in the form of rich and pleasing resonances. Speakers should not.

Speakers should respond instantly to the input signal, create an exact acoustic replica of that signal, and stop immediately when the signal stops. No speaker can do this perfectly. Speakers are supposed to reproduce only the input signal but, because they are electromechanical devices that store and release energy, they produce some sounds that are delayed replicas of the input signal and some sounds that are essentially unrelated to the input signal.

Stored energy problems arise from a variety of sources. Energy can be stored in resonating enclosure panels or other structural components. Energy can be stored in driver diaphragms. This energy will be released after a short delay, smearing the signal over time. Energy can be stored in the electrical components of the crossover network and drivers, too. Inertia will cause the mechanical parts of the drive elements to continue to move after the driving force ceases. This energy will be released in the form of ringing or oscillation after the signal stops.

Reflected Energy

Reflected energy has deleterious effects, too. Reflected energy arrives at the listener's ears after the primary signal which comes directly from the speaker diaphragms. Reflected energy has a negative impact on sound which is similar to the detrimental effects described above. It smears the signal in time, blurring definition and obscuring musical information.

Energy can be reflected from the baffles on which the drivers are mounted and from other cabinet structural components. Energy can be reflected from the driver baskets and magnets which are directly behind the radiating diaphragms. This energy will be returned to the listener through the diaphragms. Energy can be reflected from the room boundaries near the speaker.

Some speakers create reflected energy on purpose (bipoles and "direct/reflecting" designs). Whether the cause is accidental or intentional, the results are the same: a loss of detail, definition and image focus.

Artificial Ambience

Some speakers, like the Mirage and Definitive Technology bipolar models, direct as much energy towards the room boundaries as they do towards the listener. This creates an artificial ambience effect similar to what you get from "direct-reflecting" speakers like the Bose 901. Speaker designs which bounce a lot of sound off the walls offer an artificial sense of spaciousness that many neophytes find appealing. As they gain listening experience most people will learn that this spatial effect is not only false, it's achieved at the expense of detail and resolution. Energy that is reflected from the room surfaces directly behind the speakers blurs transient response and image focus. There is no such thing as a high-resolution bipolar speaker, in my opinion, because the actual resolving power of the speaker will be obscured by this reflected energy.

Dipolar planar speakers, like Magneplanar planar-magnetic models and the Sound Labs and Martin-Logan electrostatic models, direct half of their acoustic output towards the front wall (the wall behind the speakers) too. Unlike bipolar speakers, the back wave from dipolar planar speakers is out of phase from the front wave because both front and back waves are created by the same diaphragm. Dipolar speakers have virtually no radiation to the sides so there will be less energy reflected from nearby side walls than with bipoles, but the reflections from the wall behind the speaker will be just as problematic with dipolar speakers as with bipolar speakers. Dipolar planar speakers have even more serious energy storage problems inside.

The Problems with Planars

We've been discussing the problems of timing the multiple drive elements in a conventional electrodynamic loudspeaker. Wouldn't all these problems be eliminated with a full range planar speaker with no crossover network and only one drive element? Unfortunately, the answer is an emphatic no.

Planar speakers do have some attractive attributes. Since there is little or no enclosure there will be few enclosure aberrations added to the sound. No box resonances will be heard because there is no box. Few diffraction effects will alter the direct sound from the radiating diaphragms because there is little structure around the diaphragms. There will be no energy reflected from the front baffle because there generally is no front baffle. There will be fewer crossover aberrations added to the sound because, if there is a crossover, it will have only one transition point. A full range planar speaker will have no crossover network at all and a two-way planar speaker will have a simple crossover with a single transition between drive elements.

Planar speakers are free from many problems associated with conventional dynamic driver speakers. Most listeners have become accustomed to the faults of conventional speakers and when they hear a design with different faults, to which they are unaccustomed, they may be momentarily fooled into believing that planar speakers are perfect. This is far from true. Planar speakers have aberrations and performance compromises, too. I find many of these compromises to be far more objectionable than the conventional speaker faults that we all know and abhor.

Few planar speakers are 1-way designs. Most have a woofer and a tweeter and a crossover network just like their more conventional counterparts. Some 2-way designs, like the Magneplanars, have the woofer and the tweeter side-by-side. In this arrangement, tonal balance will change with a change in horizontal listening position. Listeners seated side-by-side will experience different sound because they will be at different distances from the drive elements. One listener will be closer to the tweeter and farther from the woofer, and vise versa. Vertical arrays like the Martin-Logans, solve this problem but create another one. They combine a point-source woofer with a line-source tweeter. These dissimilar drive elements will only blend in amplitude at one listening distance and the balance between the elements will be different at all other distances (see issue 2 of the Audio Perfectionist Journal). The sound reflected from the room boundaries (particularly the front wall) will have a different tonal balance than the direct sound that the listener hears. The dissimilar drive elements never do blend properly, in my opinion, so this point is not worth further elaboration, as we'll see when we look at the measured response of a Martin-Logan model.

The few speakers that actually are true 1-way designs have large radiating surfaces which must respond to full range audio signals. This creates many unique problems. Intermodulation distortion is increased because the same diaphragm surface must reproduce both low frequencies and high frequencies. Comb filtering occurs due to the many different path lengths from various parts of the large diaphragm to the listener's ears. Bass response is compromised due to dipole cancellation. Electrostatics (and some ribbon tweeters) are transformer coupled adding the aberrations of the transformer to the equation.

All large planar speakers, whether 1-way or 2-way, have diaphragms that are clamped at the edges. These diaphragms can never operate as true pistons. They actually function more like tympanic membranes. The tension on the membranes is impossible to accurately control so no two examples will be identical. These diaphragms will store and release energy for a long time after the signal stops (just like the skin on a drum), as we'll see when we examine the cumulative spectral decay plots that follow.

The curved Martin-Logan panels are inherently non-linear because diaphragm tension increases when the diaphragm moves forward and decreases when the diaphragm moves rearward. The segmented panels of the Sound Labs electrostats are more linear but interference patterns develop between segments. The flat Magneplanar bass panels are inherently non-linear because the magnet structure is only on one side. When the diaphragm moves towards the magnets the voice coil is reacting to an increasing magnetic field. When the diaphragm moves away from the magnets the voice coil reacts to a decreasing magnetic field.

I could go on and on but, after you see the frequency response measurements of these various speaker types, the futility of this will be apparent. All these planar speaker designs create an artificial sound which is demonstrably inaccurate. If you like the sound, then discussing the technical faults won't mean much.

Coloration

Coloration is a term that refers to unwanted sounds that are added to the signal. Speakers can "color" the sound due to inaccurate frequency response, inaccurate phase response, energy storage and release, reflections added to the signal and a number of other factors. Obviously, we don't want speakers that impart their own characteristics to every piece of music we play through them. So what do we want if our goal is the most accurate possible reproduction of the signal which represents the recording?

What Should Accurate Speakers do?

Accurate speakers should have flat frequency response so that all frequencies can be reproduced with equal intensity. Accurate speakers should be time- and phase-correct so that harmonics are reproduced with the proper temporal relationship to fundamentals. Accurate speakers should start and stop in step with the input signal. Accurate speakers should not add artificial reflections or other colorations to the signal.

How can we determine which speaker models are likely to be truly accurate? Are there meaningful measurements which can help us weed out the poor designs before we start listening? Can fish breath under water?

Measurements

No measurement can tell you everything about how a speaker will sound. No single measurement technique or graph can be of much value alone. A group of measurements, each compiled in exactly the same manner, can give you a strong indication of the designer's expertise and offer valuable comparisons of the performance capability of different speaker designs.

The most important speaker measurements are those that tell us about frequency response linearity, time- and phase-coherency, and energy storage problems.

A speaker with gross deviations from flat frequency response cannot accurately reproduce the input signal, no matter how appealing its colorations may be to some listeners. You can make pretty good sound with speakers which have flat frequency response yet scramble the temporal relationships between the fundamentals and the harmonics of music but you'll never get great sound, in my opinion.

Speakers that scramble phase and smear information over time cannot accurately reproduce the input signal, no matter how appealing some of these effects may be to some listeners. Speakers that scramble phase will not allow listeners to fully appreciate the new, higher-resolution recording formats like SACD which provide a more phase-coherent signal than what is available from CDs. (There will be a lot more about this in future issues of the AP Journal.)

Certain measurements can tell you all about the frequency response and phase response of a loudspeaker. Other tests can tell you a lot about energy storage problems. Sometimes useful measurements actually get published in magazines.

The measurements that Stereophile publishes with speaker reviews tell me a great deal about how well these speakers were designed and how they will probably sound, but they may not provide you with much useful information because you may not know how to interpret the graphs they print. The truth is they really don't want you to know.

Stereophile measures all the right things in (mostly) all the right ways but they don't spend much time telling you what the measurements mean. If they did, they would risk offending some big advertisers whose poorly designed speakers are often exposed by the measurements printed along with the rave reviews that all expensive, heavily-advertised products get in that magazine.

While all measurements are helpful to speaker engineers during the design process, some measurements are more useful than others to consumers. I'm going to describe the tests that I think are the most valuable to people like us who are seeking long-term musical satisfaction and who want the most for our money. In the following article we'll examine the standard speaker measurements and I'll try to describe what they indicate.

Meaningful Speaker Measurements

There are many ways to measure speakers and many different instruments are available to implement the various tests. I would caution readers that direct comparisons between different testing methods and different measuring devices are dangerous. You can compare John Atkinson's MLSSA graphs in Stereophile, one to another, but you're walking on thin ice when you try to compare his results to the results of another tester using different instruments or methods. There are many other pitfalls that I'll try to mention along the way. Remember, there is no substitute for experience and novices usually don't have any. Many magazine reviewers have little more. Atkinson knows what he's doing but there are many others who could best be described as clueless. Use test graphs as a preliminary evaluation tool only and don't try to read into them more than your experience level allows. Pay little attention to the interpretations of a reviewer unless you have come to trust that individual through experience.

I'll start by describing various measurements and then we'll discuss how you can interpret the graphs produced by these measurements.

The Wonderful Square Wave

A square wave is a wonderful thing. Technicians love square waves because they can evaluate many aspects of performance with a single test. Using just two frequencies—100Hz and 10kHz—a technician can test the frequency response of a complete amplifier, or a single stage of amplification, from 10Hz to 100,000Hz.

A square wave looks simple but it is really a complex waveform made up of many sine waves including the fundamental frequency and all the odd harmonics of that frequency. This closely mimics the structure of musical waveforms which include fundamentals and harmonics which must be properly represented for accurate reproduction. The flat top of the square wave tests the low frequency response of a circuit because it resembles low frequency signals or short bursts of DC voltage. The vertical portion of the square wave contains high harmonic frequencies that test the response of the circuit to about ten times the fundamental frequency. When the phase relationships between the harmonics and the fundamental are altered the wave changes shape and may no longer resemble a square wave at all.

Few speakers can reproduce a square wave and none can reproduce one at low frequencies. The DC portion cannot be acoustically reproduced in air because when the speaker diaphragm stops moving at the maximum points of excursion, no pressure waves are created in the air surrounding the diaphragm and acoustical output ceases. If speakers can't really reproduce square waves, why are we discussing square waves in an article about speaker measurements? Because square waves and related signals are the origin for many computerized speaker tests. The step response test is essentially half a square wave. An impulse is the same thing with a shorter period. Modern test instruments correlate an impulse from a stimulus called the Maximum Length Sequence (MLS) and I'll try to offer a brief explanation of what that is.

MLS and FFT

Modern computer-based loudspeaker measurement systems utilize the Maximum Length Sequence (MLS) stimulus and calculate all response measurements using the Fast Fourier Transform (FFT). Even if I could fully explain how these things work, that explanation would be difficult to comprehend for those of us who are not mathematicians and would be of little interest to most readers. I do understand the basic concepts well enough to use these modern, computerized instruments to obtain meaningful measurements for loudspeaker performance. I'll try to give a rudimentary explanation of the terms here and then we'll go on to practical information about how to use the results.

The MLS stimulus is a discrete number sequence that switches between two numbers in an almost random way, producing what is called "pseudo-random noise." This "noise" is not really random because it can be repeated exactly and it is periodic in nature. The MLS power spectrum is similar to white noise which contains all frequencies in a defined ratio. These facts make the MLS an ideal source for FFT analysis.

The Fourier Transform is named for Jean Baptiste Joseph Fourier who died in 1830. Daniel Bernoulli introduced the sine-cosine series in the early 1700s. The Discrete Fourier Transform (DFT) and the FFT algorithm, which allows the efficient computation of a version of the DFT, form the basis for modern digital signal processing. The FFT transforms time-domain samples into frequency-domain data.

Before you lose interest here's the bottom line: In the old days we put a speaker in a huge anechoic chamber, fed it sine waves and measured what came out with a microphone and a volt meter. Today a computer sends some funny noises to a speaker under test for a few seconds. The output from the speaker is captured by a microphone and fed back to the computer which then calculates all aspects of the speaker's performance mathematically. The computer can remove the sound contributed by the room so that measurements (down to a few hundred hertz) can be carried out in a conventional listening room. The computerized tests correlate very well with the traditional methods and the two can be combined, as we'll see from the measurements that will be presented later.

Impulse Response

Time-domain data can be converted into frequency-domain graphs by the Fourier Transform. An impulse is the primary time domain component used for speaker measurement. An impulse is a short burst of signal that contains all frequencies. An impulse has a constant, flat spectrum at all frequencies. The impulse response is the time domain equivalent of frequency response. Modern, computer-based test instruments correlate impulse response from the

MLS stimulus.

The computer can mathematically calculate a great deal of information about speaker performance from the impulse response. Transfer function, frequency response, phase response, step response, and cumulative spectral decay are all obtained from the impulse response by FFT analysis. FFT analysis is a complicated mathematical process that is performed by the computer at high speed. FFT can transform time-domain data to frequency-domain data and vice versa.

Step Response

The step response is the most useful indicator of a loudspeaker's time coherence. A step is an impulse with a slightly longer period. It's like half a square wave.

The step response test shows whether all drivers in a speaker system are pushing together to create a time coherent signal or if some drivers are pulling while others are pushing, scrambling the time relationships between the fundamentals and harmonics of music.

While the input signal for the step response test might look like the top half of a square wave, the acoustic output from the speaker can't duplicate the flat, top portion of the electrical waveform. A speaker can't reproduce the DC component in air. While the drivers can be displaced by a DC current, air movement will stop when the driver diaphragms stop. Sound will only be produced while the diaphragms are moving. There will be no sonic output when the drivers reach maximum excursion and rest there momentarily.

A step response graph from a perfectly time- and phase-correct speaker will look like a triangle, rising steeply at the left and tapering back down to zero output as time progresses.

Cumulative Spectral Decay

The Cumulative Spectral Decay plot, often called a waterfall plot because of its appearance, shows how a speaker system sings along after the song has ended and which individual frequencies or frequency ranges persist most. Remember, we want to hear only the signal from the recording, not the stored energy being returned from the speaker. The waterfall plot shows us how well the speaker under test will allow us to do that.

The waterfall plot displays a series of frequency response plots after successive periods of time have elapsed. The uppermost line is the quasi-anechoic frequency response and each successive plot below shows the output from the speaker after the signal has ceased. The scale at the left shows the diminishing output level in dB for each plot and the scale at the right shows the elapsed time after cessation of the signal.

Some instruments "auto-range" to self-adjust to the device under test. Be careful when comparing one graph to another. Always be aware of the range of amplitude shown in dB on the left and the range of time shown in milliseconds on the right. Try to compare apples to apples.

Interpreting the Measurements

We want speakers with flat frequency response and we can evaluate that aspect of performance by examining the quasi-anechoic frequency response graphs produced by computerized measurement instruments like the industry-standard MLSSA system by DRA Labs. We want speakers that are time- and phase-accurate and we can evaluate that aspect of performance by examining the impulse response and step response graphs. We want speakers that won't obscure musical information by singing along after the song has ended and we can evaluate that aspect of performance by examining the cumulative spectral decay plots and the impulse response graphs.

I think that the easiest way to discuss how to interpret these measurements is to show the response graphs of some popular high-end speaker systems and then talk about what the graphs tell us.

Interpreting Speaker Response Tests

A speaker response test graph shows how the speaker responds acoustically to the electrical input signal. In order to interpret these graphs you need to know the nature of the input signal and you need to know what to expect in response.

The quasi-anechoic frequency response graphs you've seen in magazines are all spliced together from two or more measurements and most are "smoothed." Quasi-anechoic testing is accomplished in a reverberant environment (like a normal listening room) by "gating" the sound that is captured by the test microphone and sent to the computer. The computer is told to ignore those sounds which arrive before the signal from the speaker reaches the microphone and to stop accepting sounds before the first room reflections arrive at the microphone. The first room reflection will generally come from the floor which is usually the reflective surface closest to the speaker. The gate is open just long enough to capture the complete signal from the speaker (at frequencies above 500Hz or so, depending on the height of the tweeter and microphone), and not long enough to include reflected sounds from the room boundaries.

Since the computer can only analyze signals between the first arrival from the speaker and the first reflected arrival from the room, low frequencies cannot be accurately measured. Quasi-anechoic measurements, like the ones that John Atkinson makes for Stereophile Magazine with a 50-inch distance between the speaker under test and the microphone, are only accurate down to perhaps 500Hz. The bass range below that frequency is measured by placing the microphone very close to the woofer, and the low frequency and higher frequency ranges are spliced together to form a graph. The bass range may include another splice if there is a port or passive radiator which must be measured separately from the active woofer. Ports and passive radiators are also measured by placing the microphone very close to the source.

The measurements we're going to discuss here were made in an anechoic environment with a microphone distance of ten feet. The speakers under test were elevated off the floor to increase the path length of the first reflected sound. The resulting measurements are accurate down to about 200Hz and that's all I'm going to show you. You've already seen bass measurements in the subwoofer articles in earlier issues and this article will concentrate on the rest of the spectrum, from about 200Hz to 20,000Hz.

The measurements shown here are more representative of actual speaker performance because they were made at a realistic listening distance of ten feet and they are not "smoothed." Smoothing irons out all the little ripples and makes broad peaks and dips easier to spot but it also disguises the small anomalies that may represent serious resonance problems. Placing the test microphone farther from the speaker also allows the outputs from the individual drivers in larger speaker arrays to properly coalesce as they would in actual use.

We'll start with frequency response graphs of five popular speaker models. These graphs were produced with the industry standard MLSSA measurement system using an ACO laboratory-grade microphone.

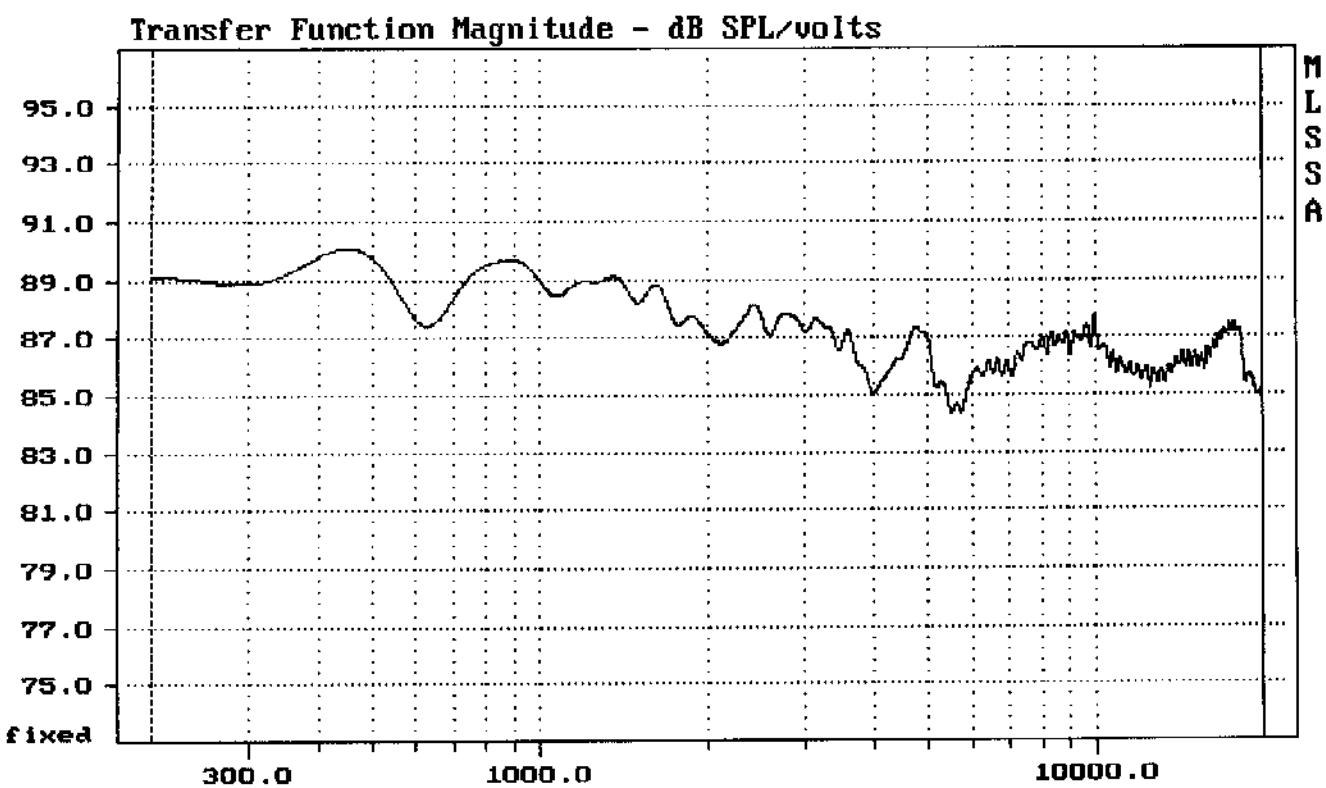
The B&W 801 Matrix speakers are a three-way dynamic driver design with "aligned" drivers and steep-slope crossover networks. The Martin-Logan Aerius speakers are a two-way electrostatic hybrid design with a dynamic woofer and a wide-range, curved electrostatic midrange/tweeter panel. The Magneplanar 1.4 speakers are a flat-panel, planar-magnetic two-way design. The Sound Lab A-6 speakers are a two-way, full range electrostatic design with a compound electrostatic woofer and an electrostatic tweeter. The Kharma Ceramique speakers are a three-way dynamic design with "aligned" drivers and a steep-slope crossover network.

At one time or another during my career in audio, I have been an authorized dealer for each of these brands except Kharma. I have sold, installed and repaired various models from all these manufacturers (except Kharma). I have done business with each of these companies (except Kharma which I know little about) and each is a well established, reputable firm. My intension here is not to disparage their products. If you like the sound from one of these products you are not alone, but this is an article about speaker measurements and accuracy. These speakers don't measure very well. The speakers we'll discuss in the next issue do produce accurate measurements. Issue 6 will allow you to contrast the two groups.

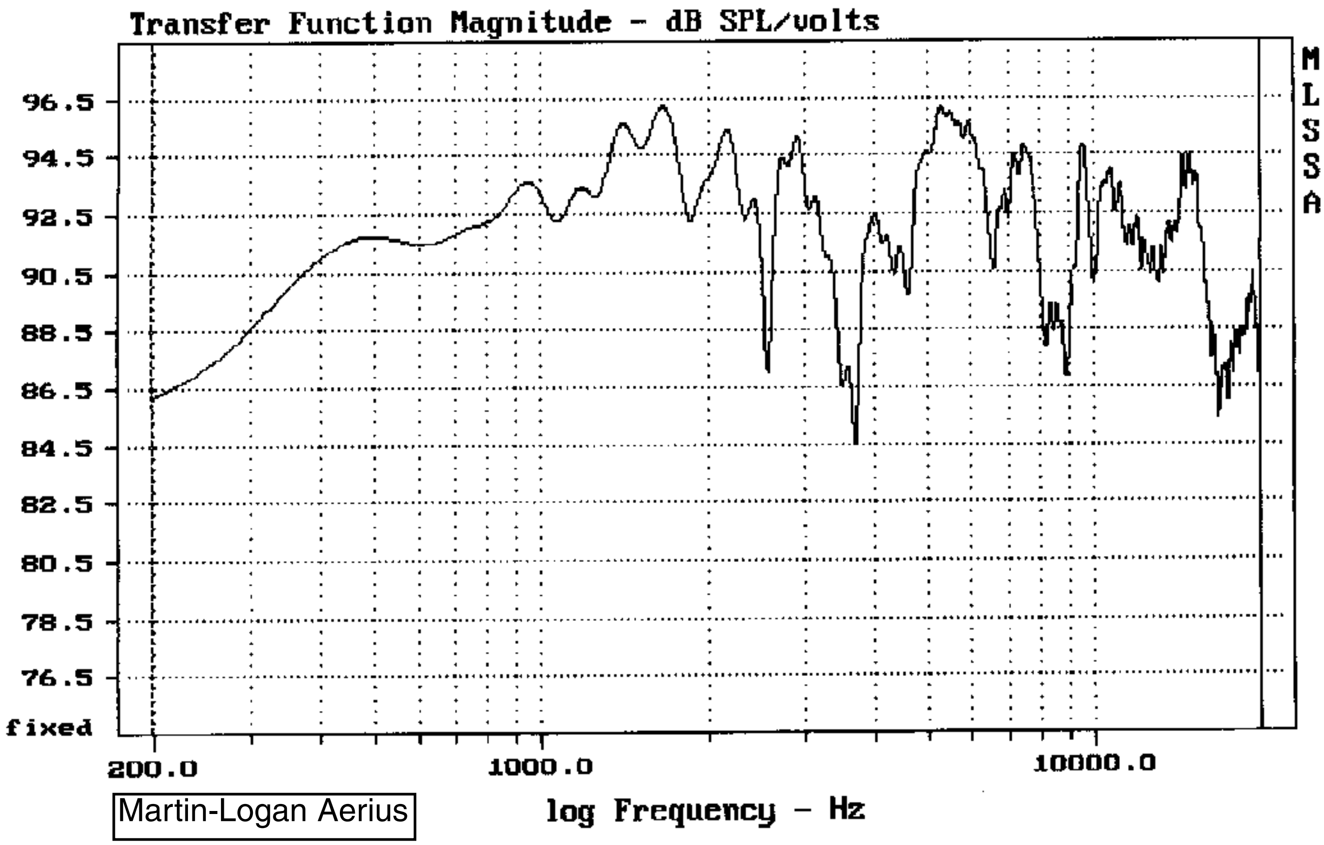
The following frequency response graphs show the output level (amplitude) in 2dB increments on the vertical scale (numbered at the left) and log frequency in hertz (numbered across the bottom). A logarithmic frequency scale has an equal horizontal distance between octaves. The distance between 200Hz and 400Hz (one octave) is equal to the distance between 3,000Hz and 6,000 Hz (also one octave).

A speaker with accurate frequency response should produce a line on the graph that stays within a range of ± 3dB (on the vertical scale) in output level. You can determine the frequency of a deviation by following the vertical lines down to the frequency scale. You can determine the amount of deviation by following the horizontal lines across to the amplitude scale and comparing the amplitude at one frequency to the amplitude of another frequency (deviations are usually compared to the output level at 1,000Hz).

Let's look at some measurements of real speakers and then I'll tell you what I see when I look at these measurements. After a few tries you should begin to get a feel for what's going on here.



B&W Matrix 801

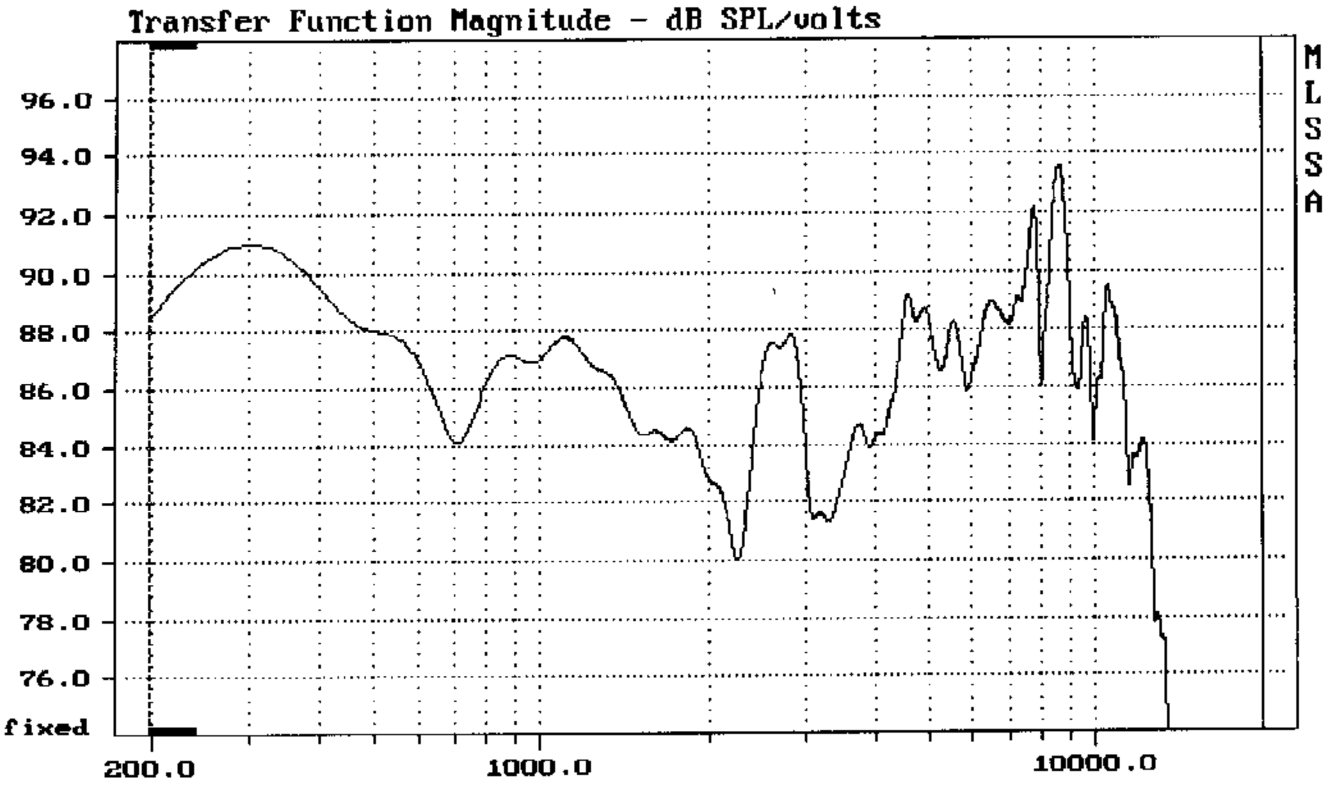


Frequency Response Graphs

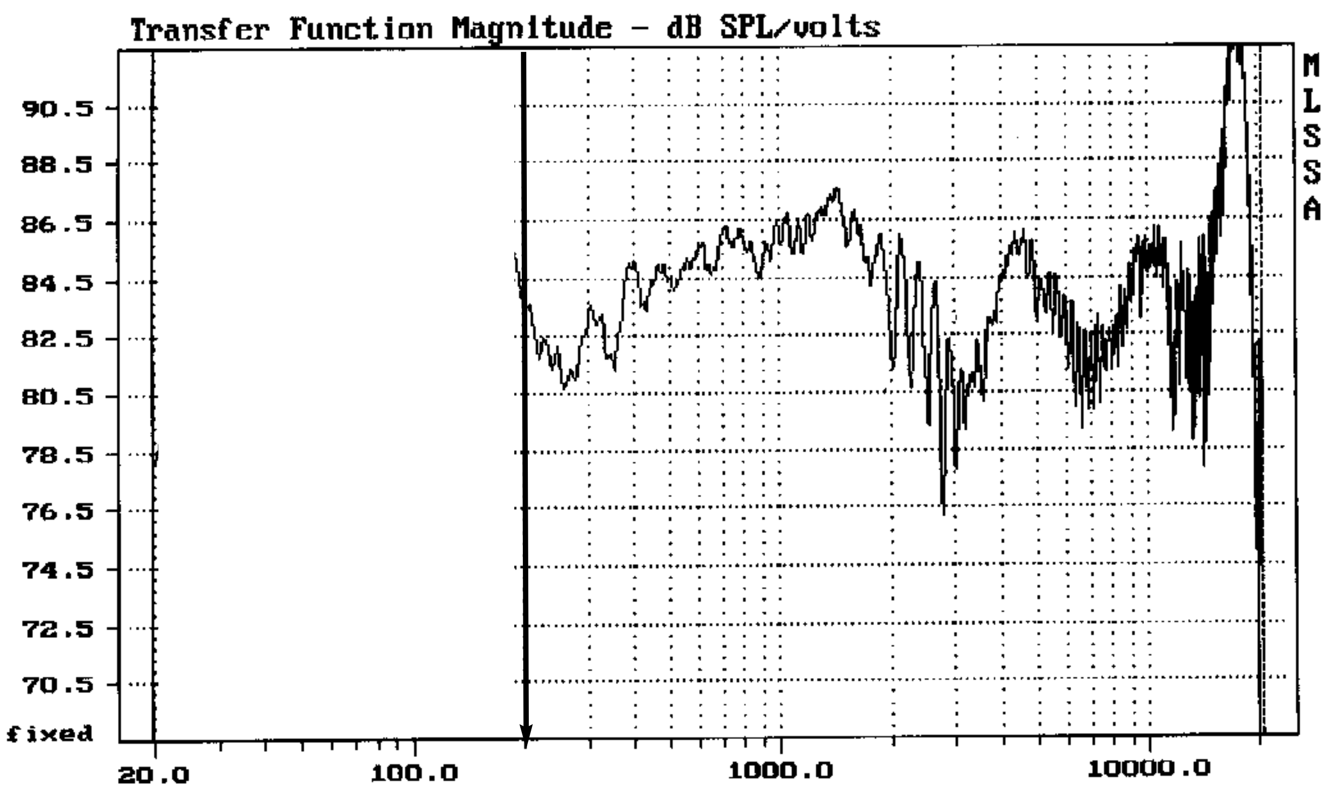
The B&W speaker has reasonably accurate frequency response within a 5dB window (approximately). Relative to 1kHz, the output from the B&W speaker varies from +1dB at 450Hz to -4.3dB at 5,600Hz. There are some minor swings between 350Hz and 700Hz which roughly encompasses the crossover point between the woofer and midrange drivers, and between 3,500Hz and 6,000Hz which probably includes the midrange-to-tweeter crossover point. This suggests that the output from the individual drivers will not blend together as smoothly as it could. Frequencies above 2,000Hz are shelved down about -2dB relative to frequencies below 1,000Hz suggesting that this speaker will sound warm, full and soft compared to a speaker with more accurate frequency balance.

The Martin-Logan speaker has remarkably poor frequency response linearity. At 1,000Hz the output is about 93dB. At 200Hz output is only 86dB, a -7dB variation. At 3,700Hz output is down to 84.5dB, or -8.5dB. At 1,700Hz output is up to 96dB or +3dB. The overall response falls within +3dB, -8.5dB for a response error window of 11dB, with huge swings in output level over the full range of the electrostatic panel! The gargantuan peak between 1,500 and 1,700Hz may sound like “electrostatic clarity” to some folks but this is an inaccurate speaker by any objective standard.

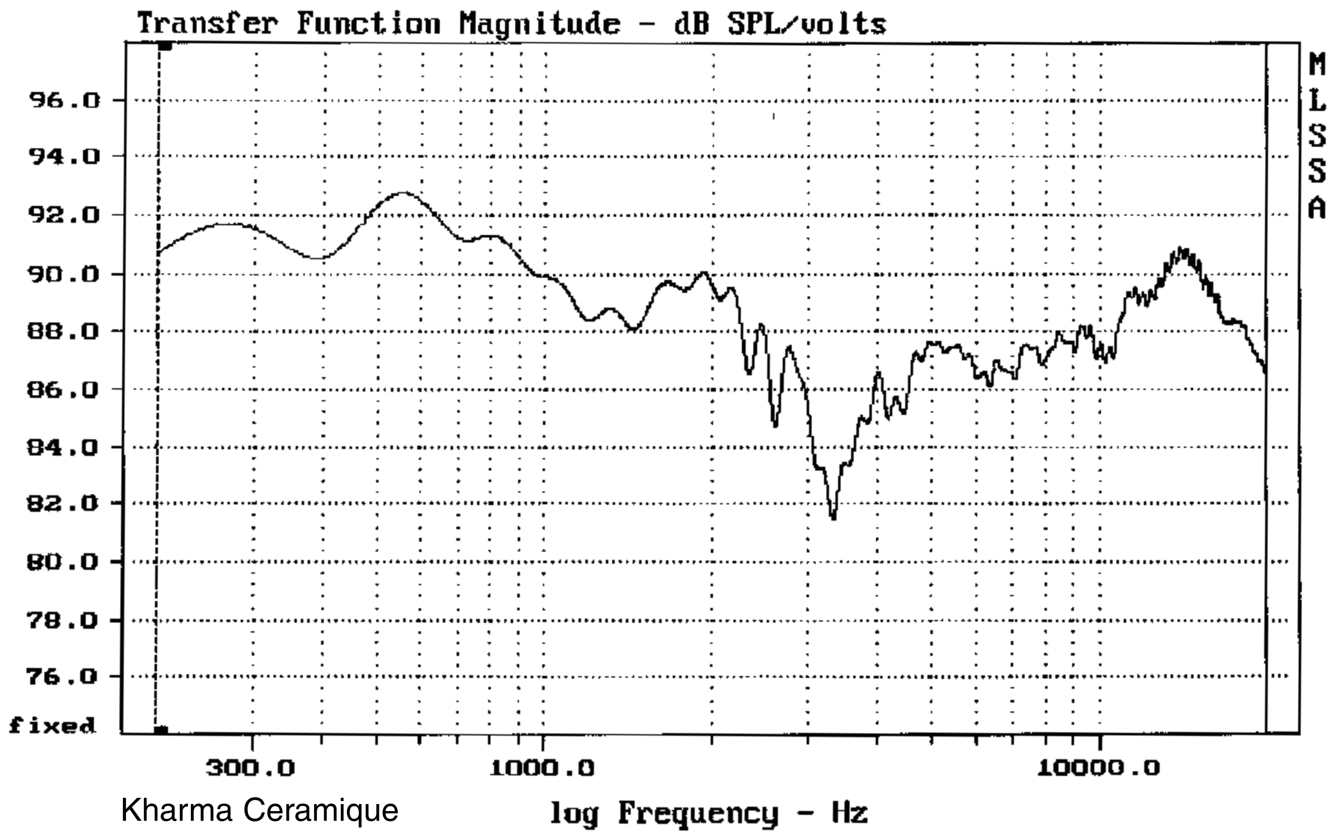
The Magneplanar speaker has a 13dB response error window but with a completely different balance. The midrange frequencies between 1,500 and 2,200Hz, which are strongly emphasized by the Martin-Logan speaker, are deeply attenuated by the Magneplanar. The Martin-Logan will probably sound “in-your-face” and highly “detailed” (over a limited range of frequencies) while the Magneplanar will probably sound smoother, warmer, and more distant due to the recessed upper midrange and the +4dB emphasis



Magneplanar 1.4



Sound Lab A-6



Kharma Ceramique

around 300Hz which will add warmth and body to the lower midrange. The large peaks in response between 8,000 and 9,000Hz will make the Magneplanar sound "airy" even though it has virtually no output above 13,000Hz. This Magneplanar model may create a lush, romantic sound that will please some listeners on some musical selections but the output signal will certainly not be an accurate reproduction of the input signal.

The Sound Lab electrostatic speaker response was plotted on a different scale so it is difficult to make a direct comparison but, as you can see, there is little need for comparison or elaboration. This is a demonstrably inaccurate speaker which emphasizes frequencies in the range where human hearing is most sensitive. It's that "electrostatic detail" thing again with even more emphasis on certain frequencies. The ziggy-zaggy response curve is probably the result of the tympanic energy storage in the stretched diaphragm or possibly from the electronics that couple the diaphragm to the amplifier. These effects may be responsible for convincing some listeners that planar speakers "retrieve more ambience" from recordings. In actuality, the speakers are creating this ambience artificially by storing and releasing energy from the stretched, edge-clamped diaphragm (and possibly the internal electronic components). Additional artificial spatial effects are produced by the dipole radiation pattern which bounces energy off the front wall behind the speaker.

The frequency response measurements from the Kharma Ceramique speaker demonstrate that conventional, dynamic speakers can deviate as far from flat response as planar speakers if their designers try hard enough (or are sufficiently incompetent). A response error window of 10.5dB is pretty bad for a dynamic speaker in modern times, but what do you expect for just a little more than $10,000 a pair? The nearly 9dB suck-out at 3,300Hz suggests a poorly designed crossover between the midrange and tweeter and the big bump centered at 550Hz suggests problems with the woofer-to-midrange crossover, too.

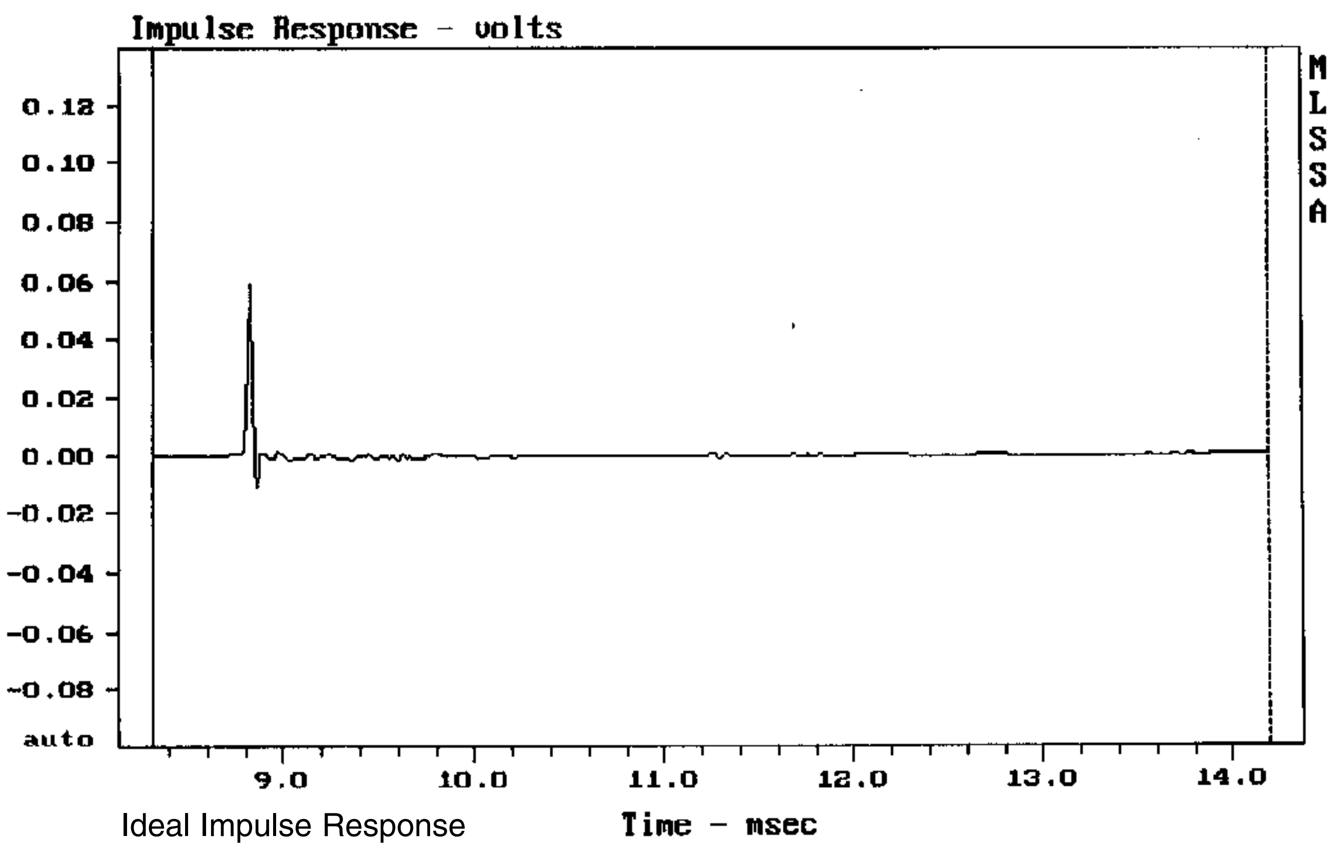
Oh well, maybe their $70,000 a pair model (which uses the same midrange driver) is better.

The response peak at about 15,000Hz may be due to a resonance in the ceramic midrange driver diaphragm or it could be caused by a breakup of the tweeter diaphragm. (The reason for a peak like this cannot be fully explained without additional measurements on the individual drive units, so don't jump to conclusions as you try to interpret these graphs.)

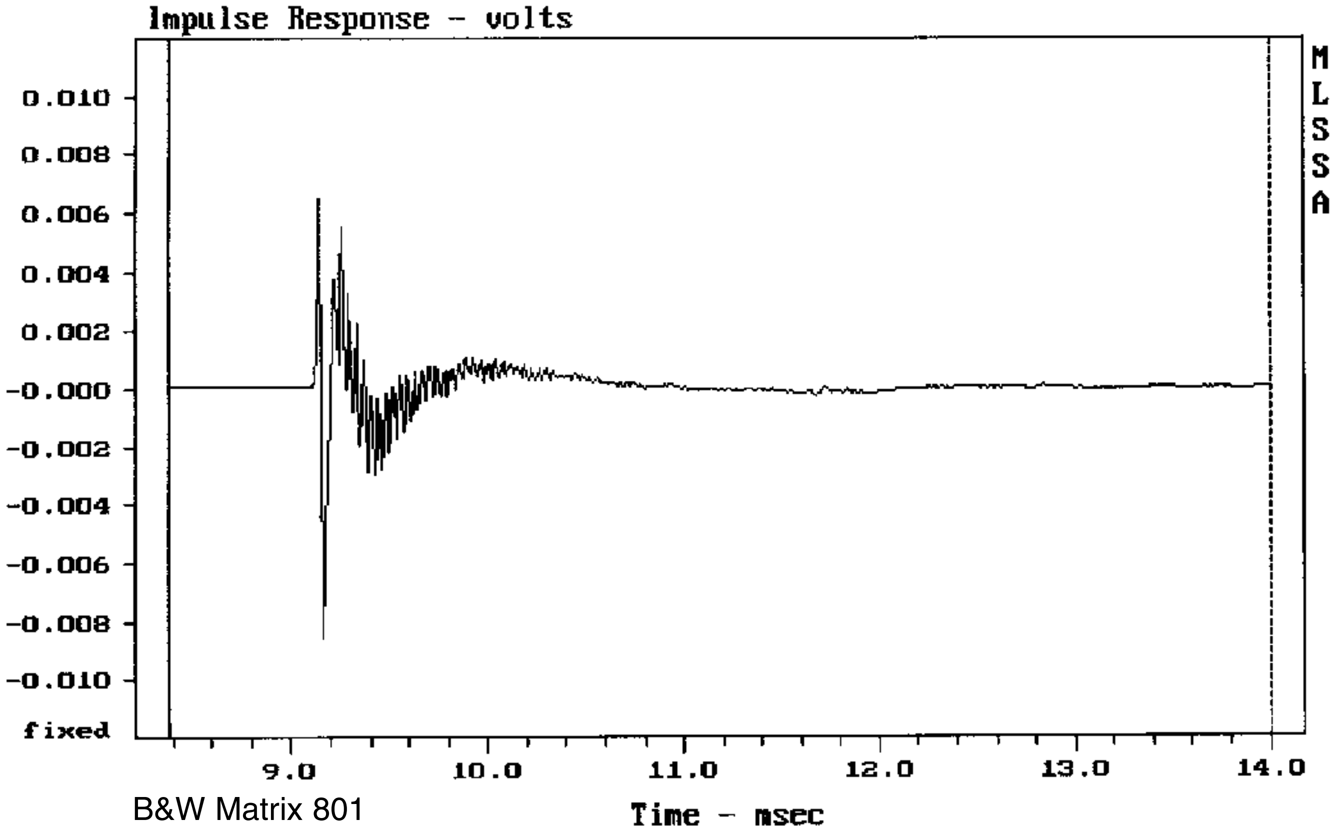
Impulse Response Graphs

The impulse response graphs show the acoustic output from the speaker under test in response to a positive electrical impulse of short duration. (These impulses are actually derived from the MLS stimulus but that's irrelevant for our purposes here.) Note that the electrical impulse goes positive only. There is no negative component to the input signal. The moving elements in the speaker drivers have mass, and therefore inertia, so the acoustical output from the speaker will overshoot the starting line somewhat in the real world. While the speaker diaphragms should only move outward in response to the positive electrical signal, inertia will cause the diaphragms to move past the starting point before coming to rest, after the signal ceases. An output that overshoots the starting line a lot, or one that rings instead of coming to rest, is indicative of mechanical or electrical energy storage. Anything other than a brief upward spike followed by a small overshoot indicates time smear. A smeared impulse response shows that the speaker under test can't start and stop in sync with the electrical input signal.

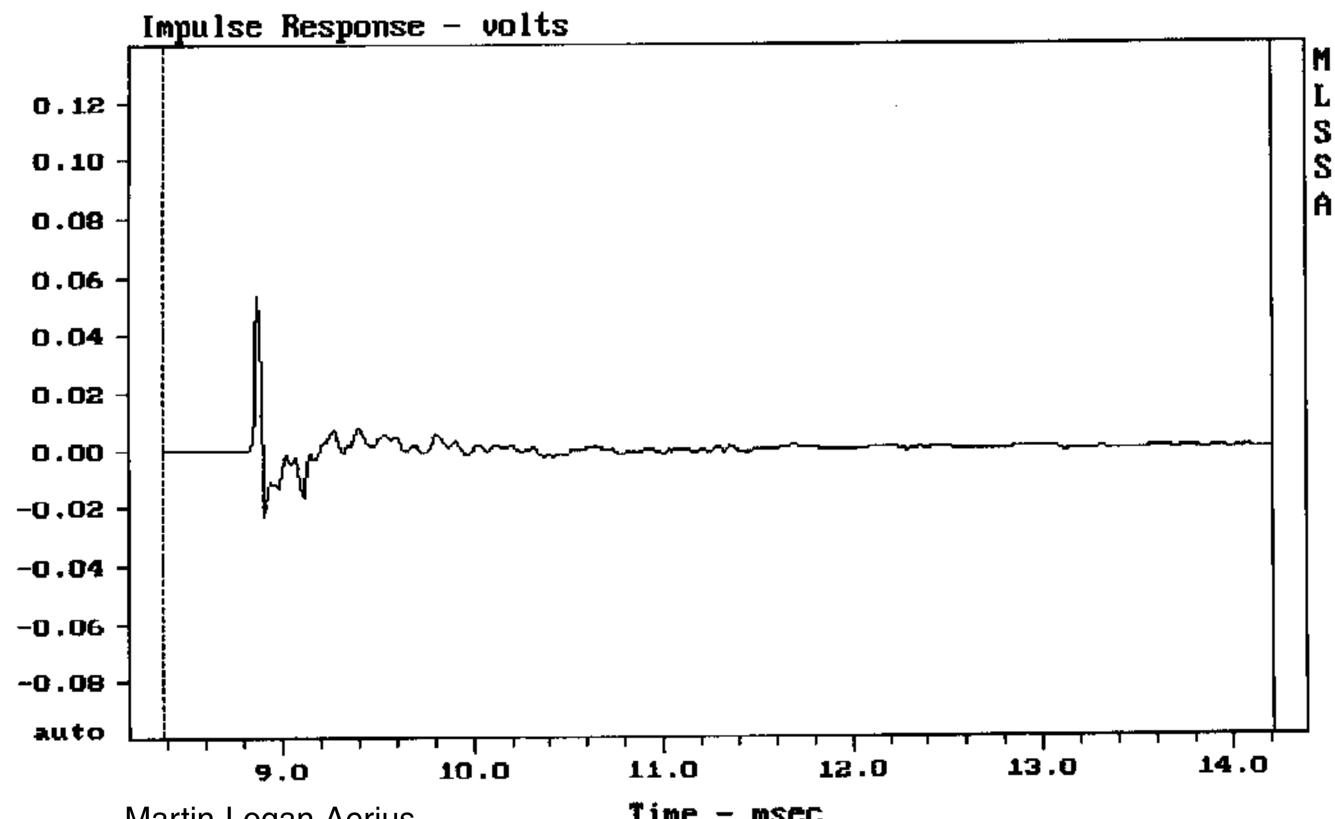
The first graph shows the impulse response from an ideal speaker. You can compare this first example to the impulse response graphs from the B&W speaker, the Martin-Logan speaker, the Magneplanar speaker, the Sound Labs speaker, and the Kharma Ceramique.



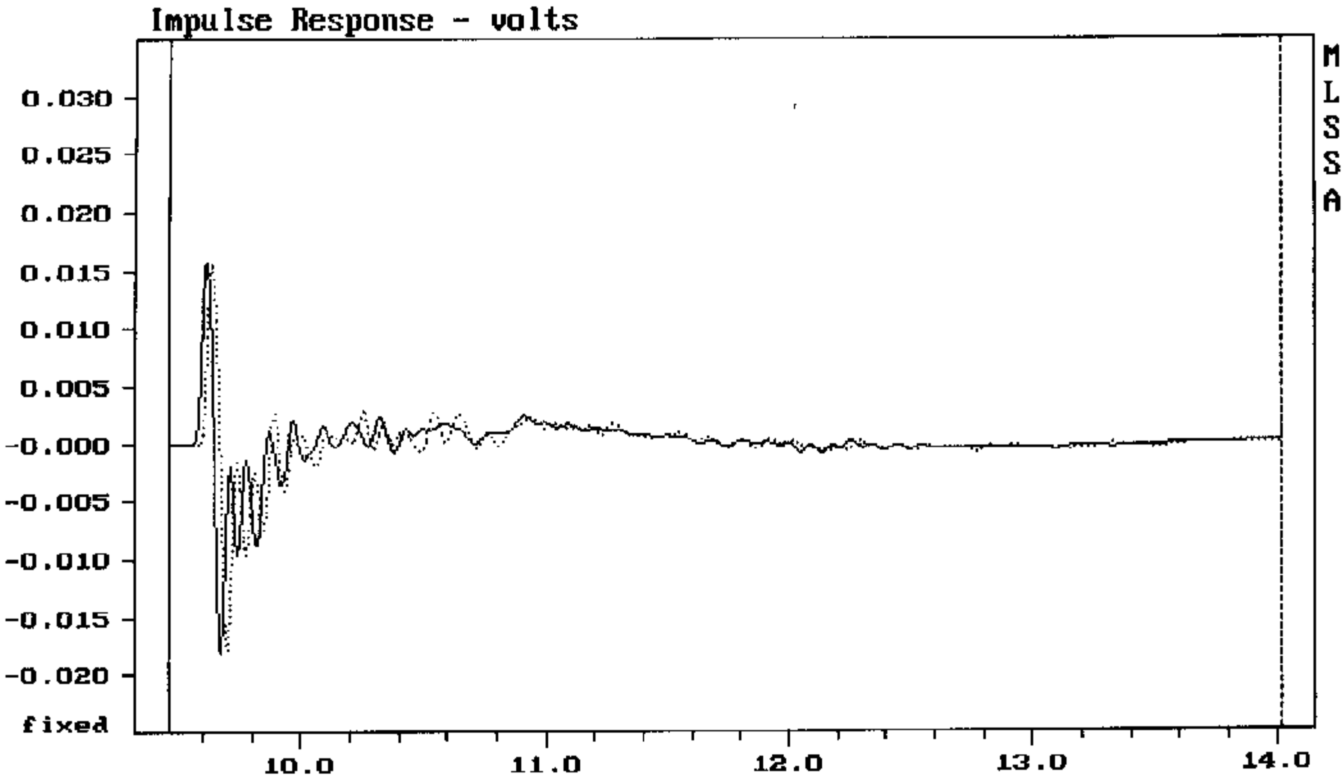
Ideal Impulse Response



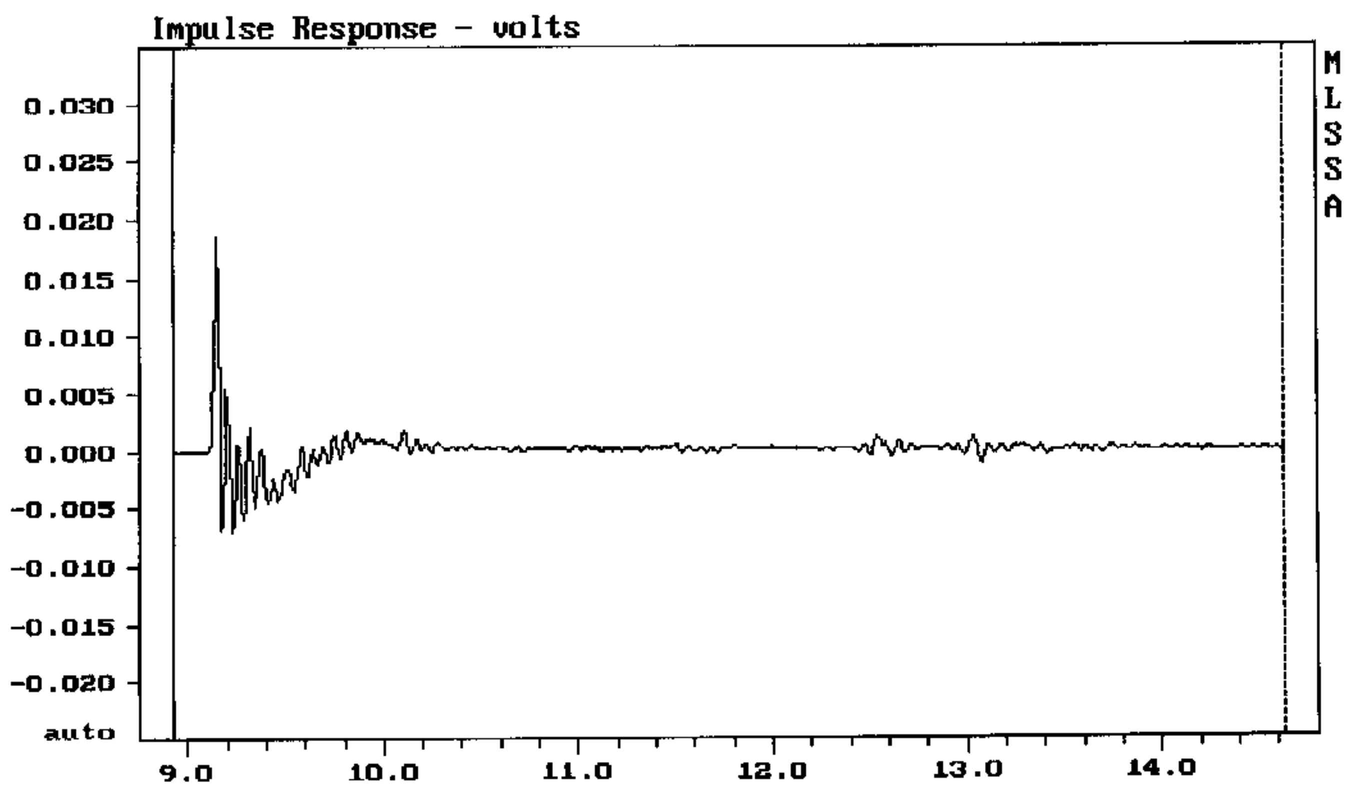
B&W Matrix 801



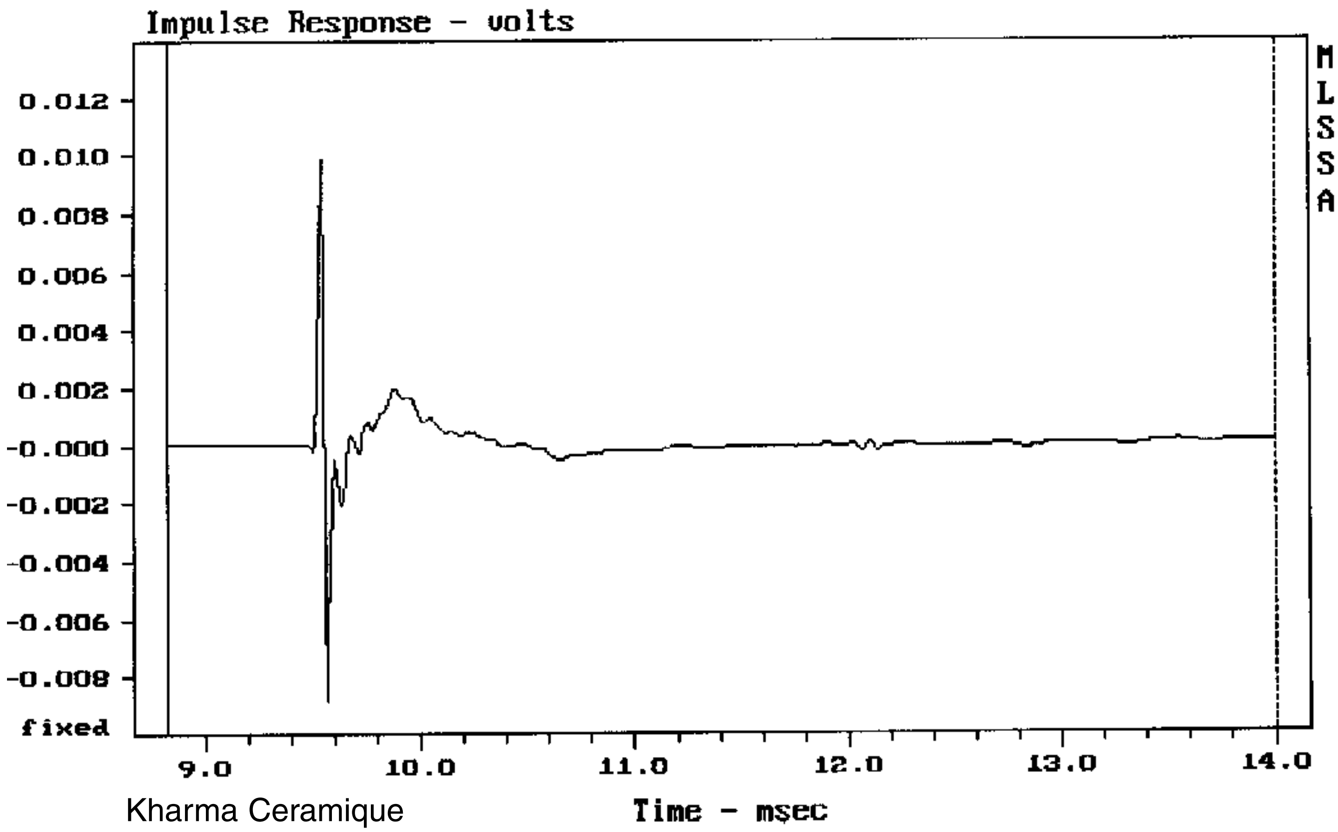
Martin-Logan Aerius



Magneplanar 1.4 (A&B)



Sound Lab A-6



Kharma Ceramique

All these speakers smear the signal over an extended period of time, as indicated by the elapsed time scale (in milliseconds) across the bottom of the impulse response graphs. (Ignore the numbers on the vertical scale at the left of the graphs.) Let's see how they respond to a step signal.

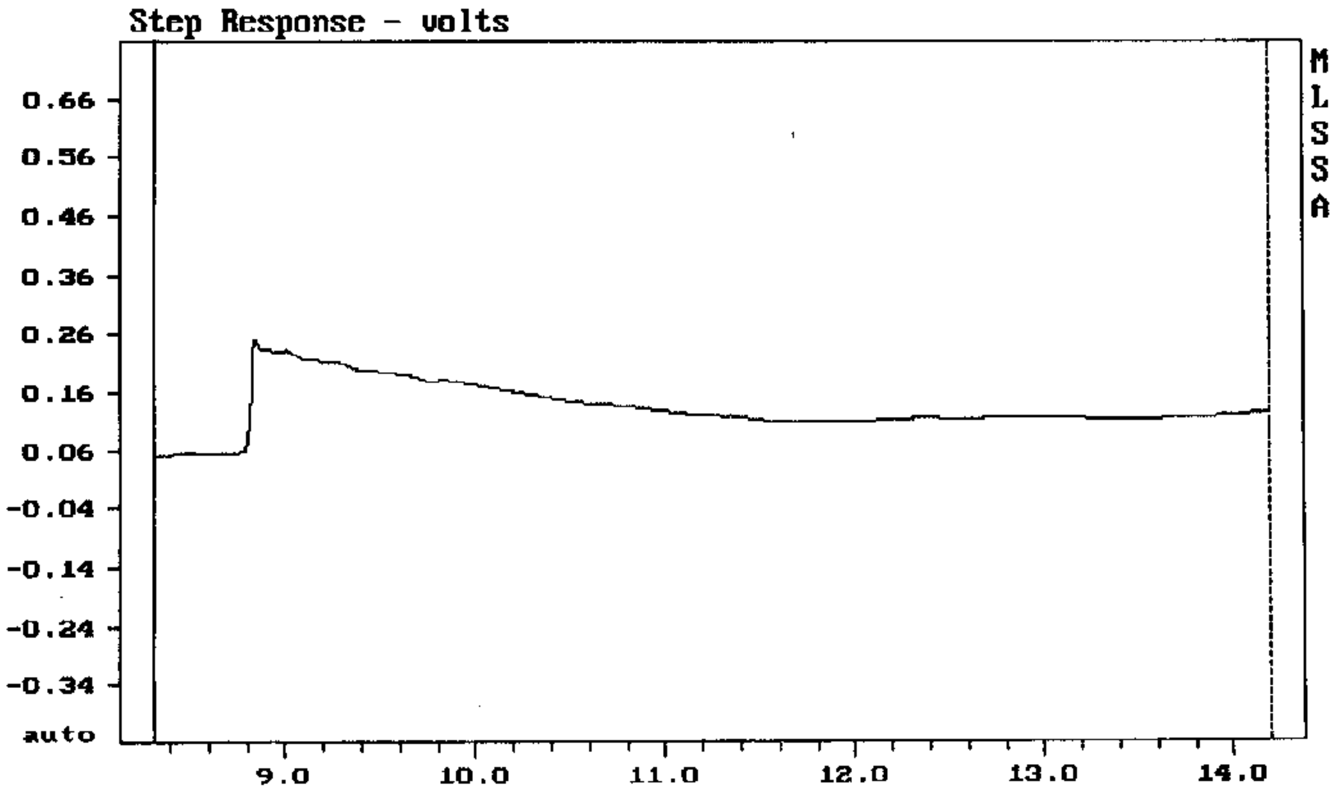
Step Response Graphs

The step response graphs, like the impulse response graphs, show how the speaker under test performs in the time-domain. Think of the electrical input signal for the step response test as the positive going half of a square wave. (These steps were actually derived from the MLS stimulus but that is irrelevant for our purposes here.)

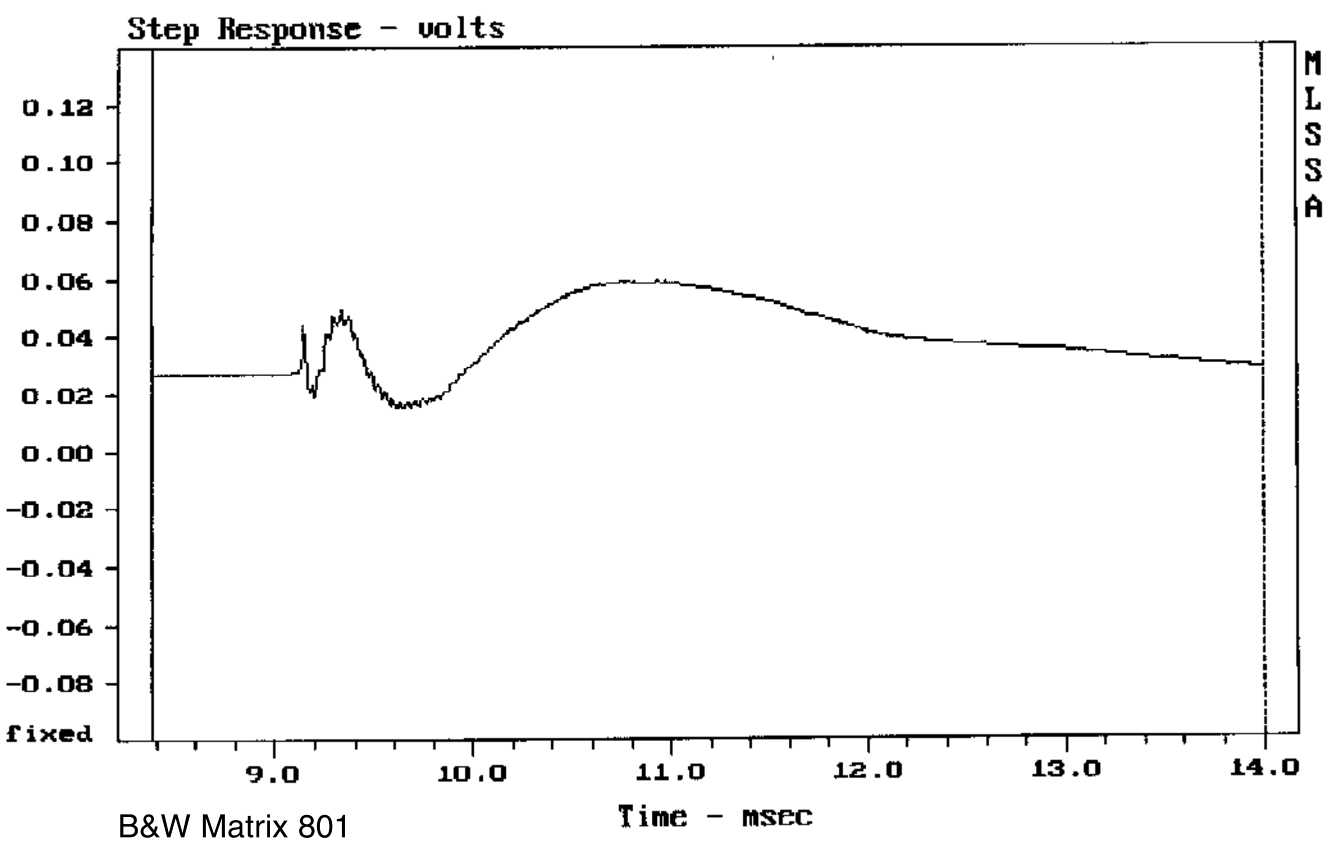
There is no negative component to the input signal. The speaker should produce positive air pressure in response to a positive electrical input. If the acoustical response from the speaker goes negative (below the starting line) it shows that some parts of the speaker are sucking (producing negative pressure) when they should be blowing (producing positive pressure). This can add new meaning to the phrase "that speaker really sucks."

Remember that musical notes include fundamental frequencies and harmonic overtones at higher frequencies. The amplitude relationships between the fundamentals and harmonics must be maintained for accurate reproduction, as demonstrated by the frequency response graphs. The timing and phase relationships between the fundamentals and harmonics must be maintained for accurate reproduction, as demonstrated by the impulse and step response graphs.

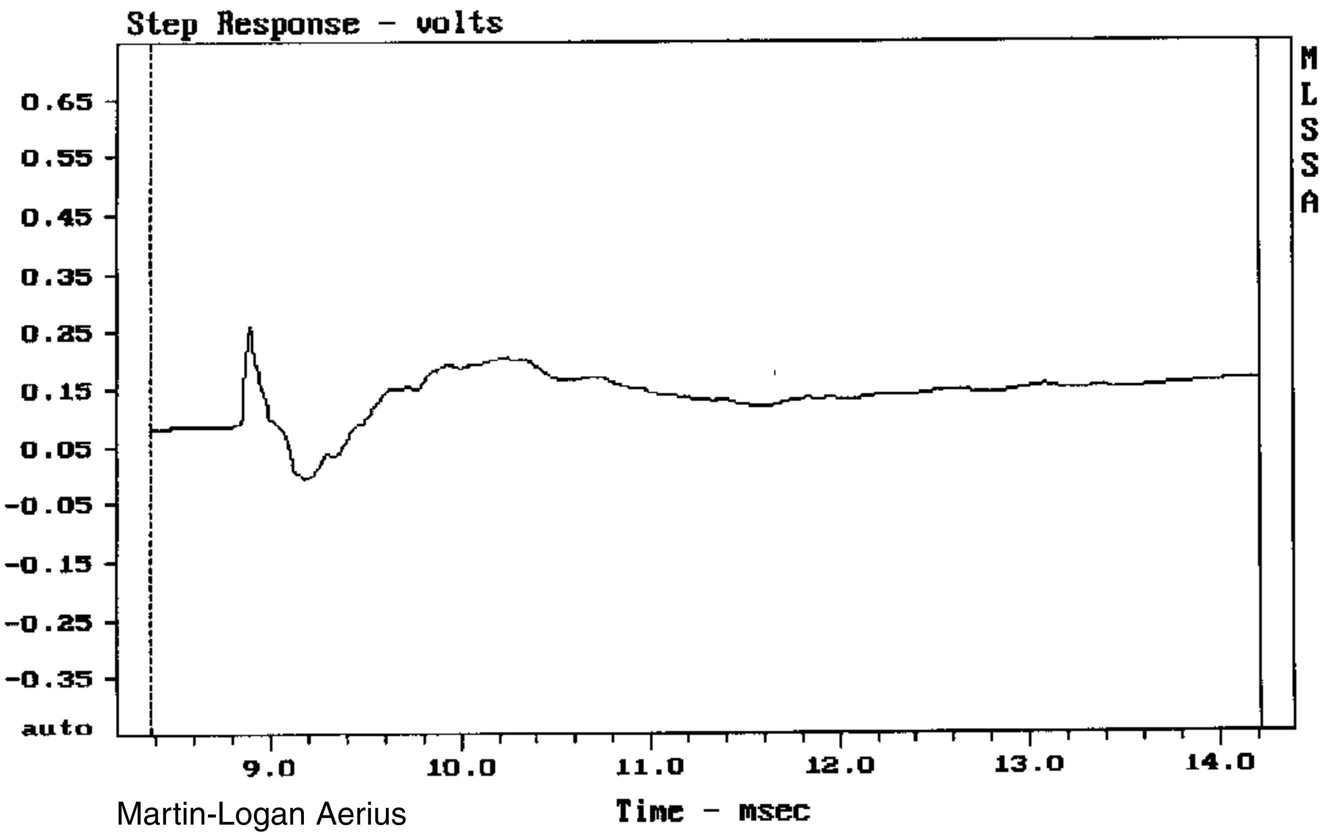
An ideal step response should be triangular in shape because a speaker cannot produce the DC component of the square wave in air. The step should rise steeply at the left and gradually slope back down to the starting line with time. A step response graph from an ideal time- and phase-accurate speaker is shown in the first illustration.



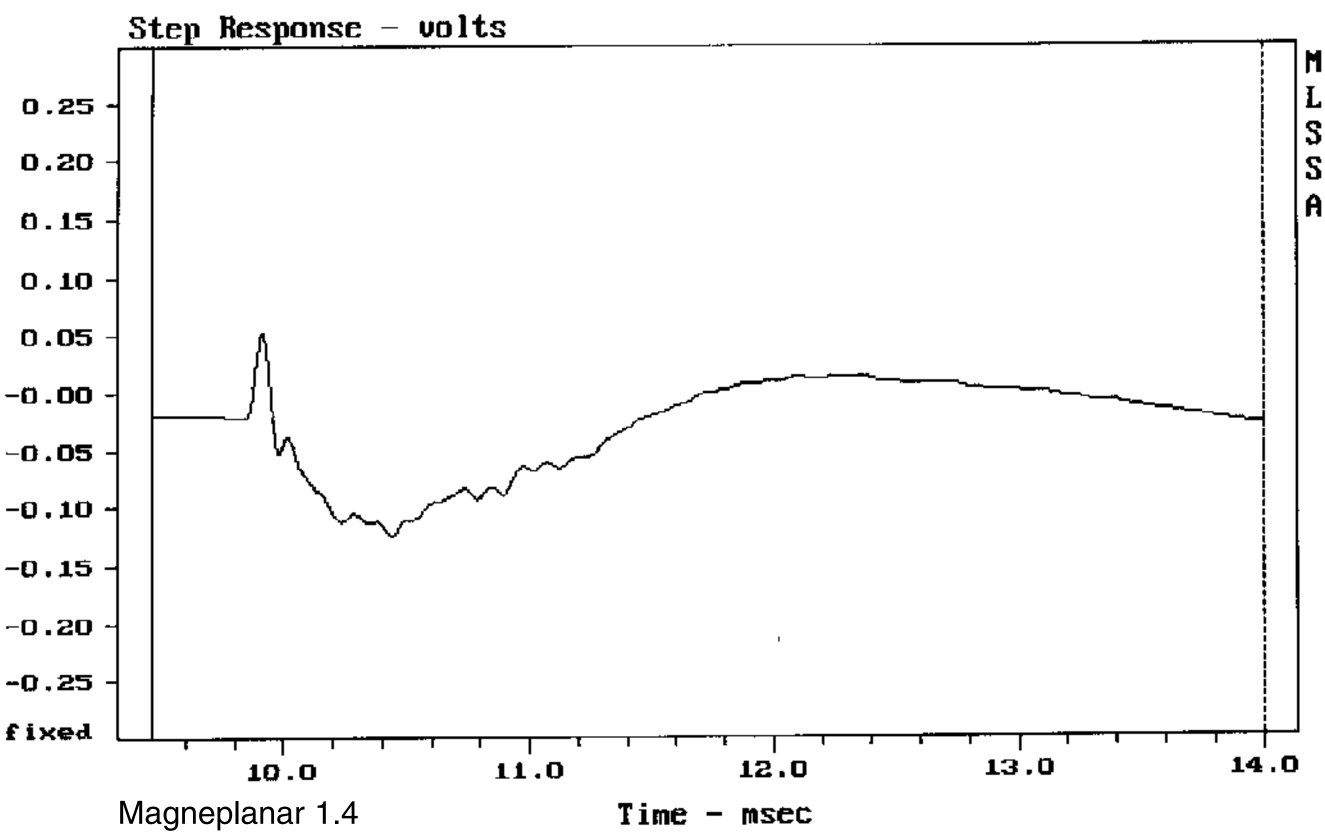
Ideal Step Response



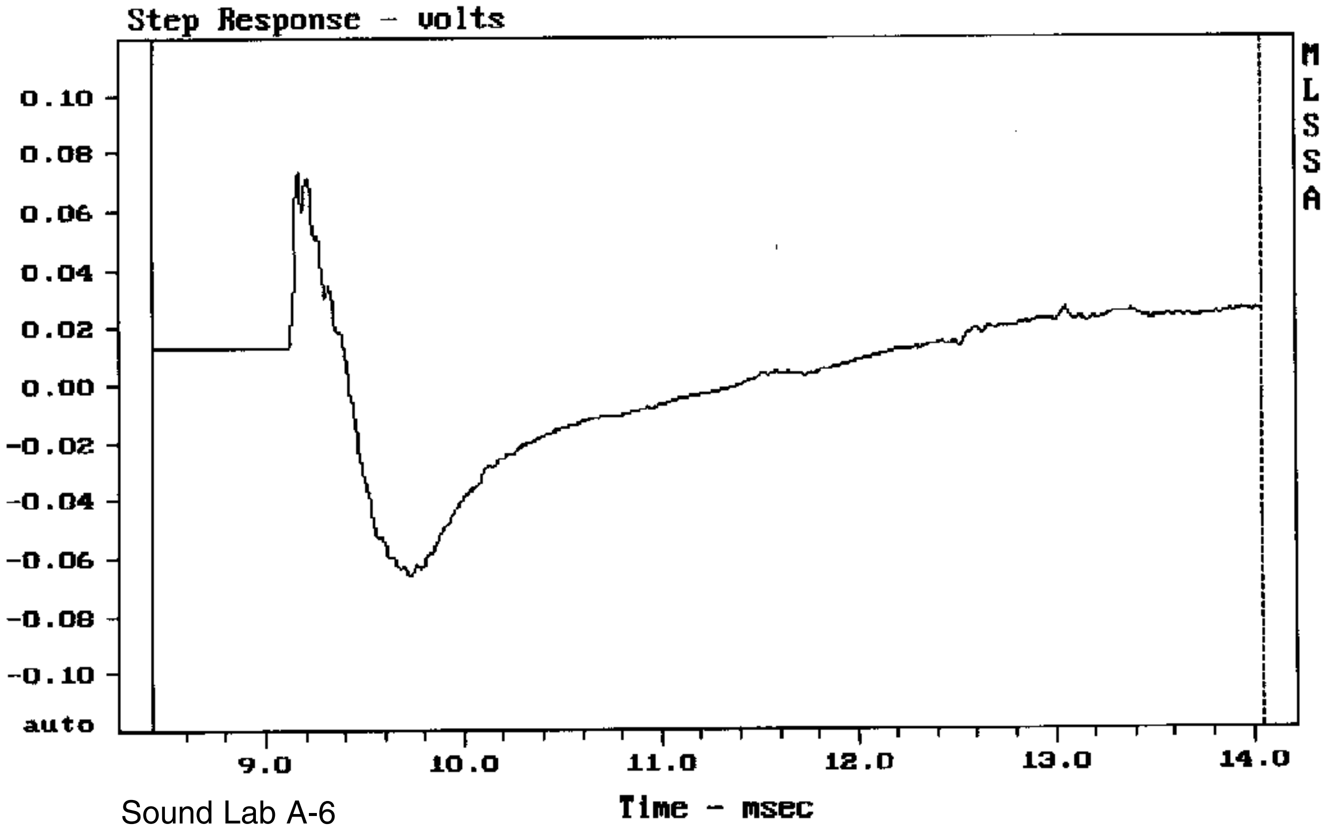
B&W Matrix 801



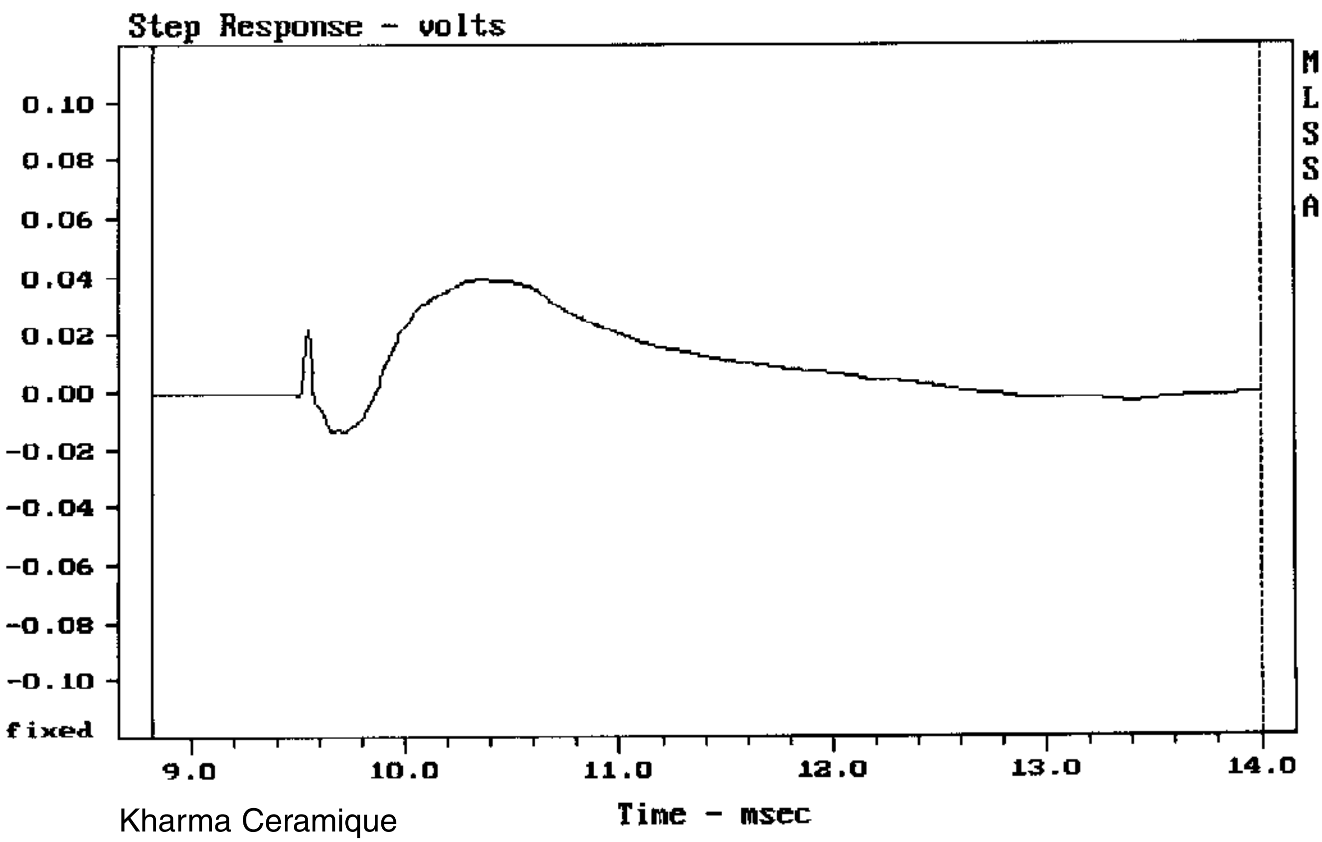
Martin-Logan Aerius



Magneplanar 1.4



Sound Lab A-6



Kharma Ceramique

You can compare the step response graphs from the other speakers to this first graph to see that not one of the speakers in this group is time- and phase-correct.

A novice can determine whether a speaker is time- and phase-coherent by looking at the step response graph of that speaker. If the graph looks like the first example, the speaker is coherent and if the graph looks like any of the other example graphs, it isn't.

Interpreting *why* a speaker produces an incoherent step response is difficult. In general, the first spike at the left will be the output from the tweeter. A broad hump occurring later will usually be the output from the woofer. If this hump is below the starting line (the horizontal line before the step which represents no signal) the woofer may be wired out of phase (connected with reverse polarity) with the tweeter but this is not necessarily the case. The woofer's acoustic output is out of phase with the tweeter if the pressure goes negative regardless of the way in which it is electrically connected. A substantial rise or fall in acoustic pressure between the tweeter spike and the woofer hump is probably related to midrange output and may come from a midrange driver if there is one, or it may indicate that the output from the woofer and/or the tweeter goes out of phase for portions of the midrange.

Cumulative Spectral Decay Graphs

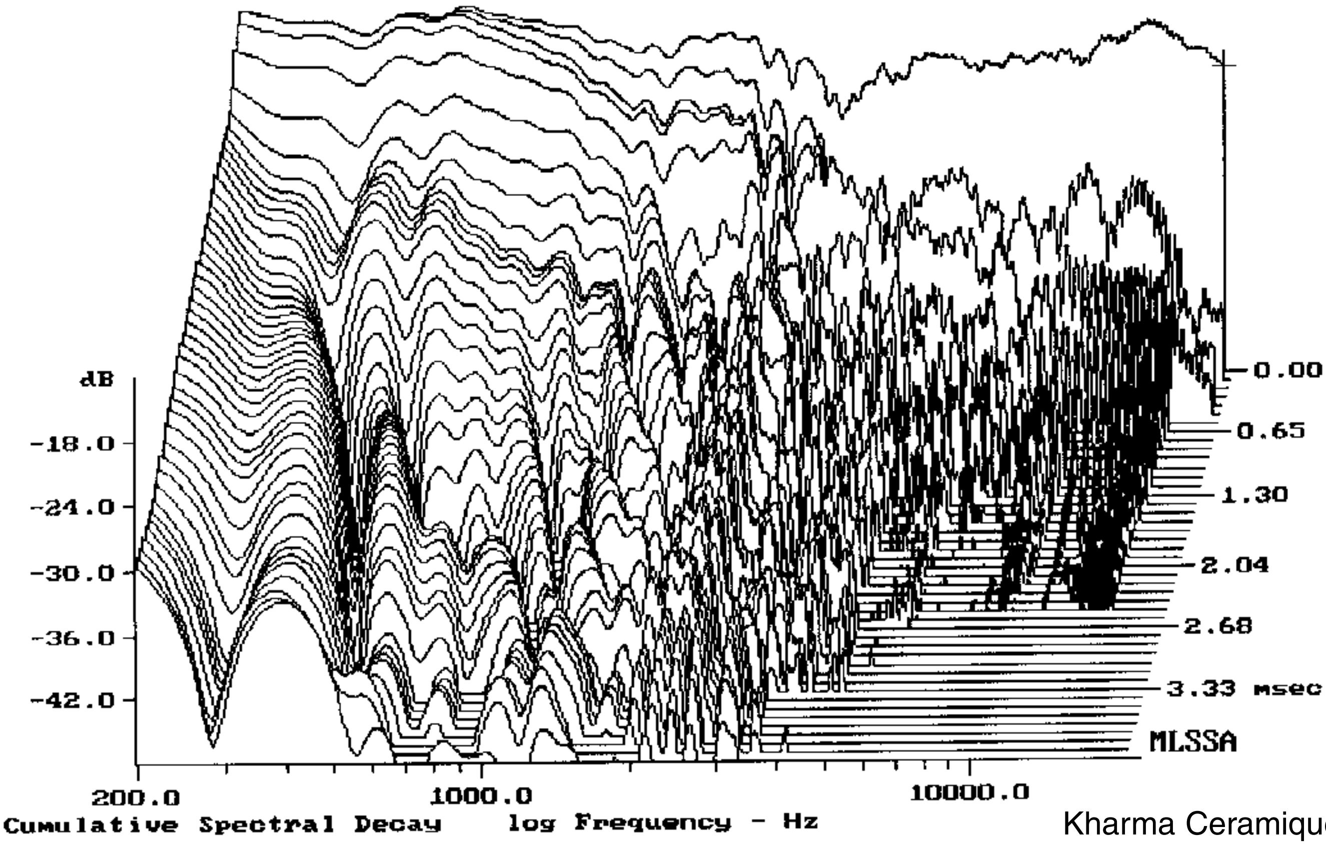
The cumulative spectral decay graphs, often referred to as waterfall plots because of their appearance, show the acoustic output from the speaker under test after the input signal has ceased. The graph shows how long the speaker keeps singing after the song has ended and how loud this delayed voice is, relative to the original tune.

Cumulative spectral decay graphs are three-dimensional plots. The horizontal line at the very top is the quasi anechoic frequency response of the speaker under test (as calculated from the impulse response). Each successive line, moving towards the bottom or foreground, shows the output from the speaker with the passing of time, after the signal has ceased. The frequency spectrum is shown across the bottom of the graph. The diminishing amplitude of the output (in dB) is shown on the vertical scale at the left and the elapsed time (in milliseconds) is shown on the vertical scale at the right. Using the instrument, the tester can place a cursor on a point on the graph to read the exact numbers on the scales. You can't do that when interpreting the graphs so you'll have to guesstimate the values. The MLSSA instrument "auto ranges" the scales so be careful to note the values on each scale when comparing one test to another.

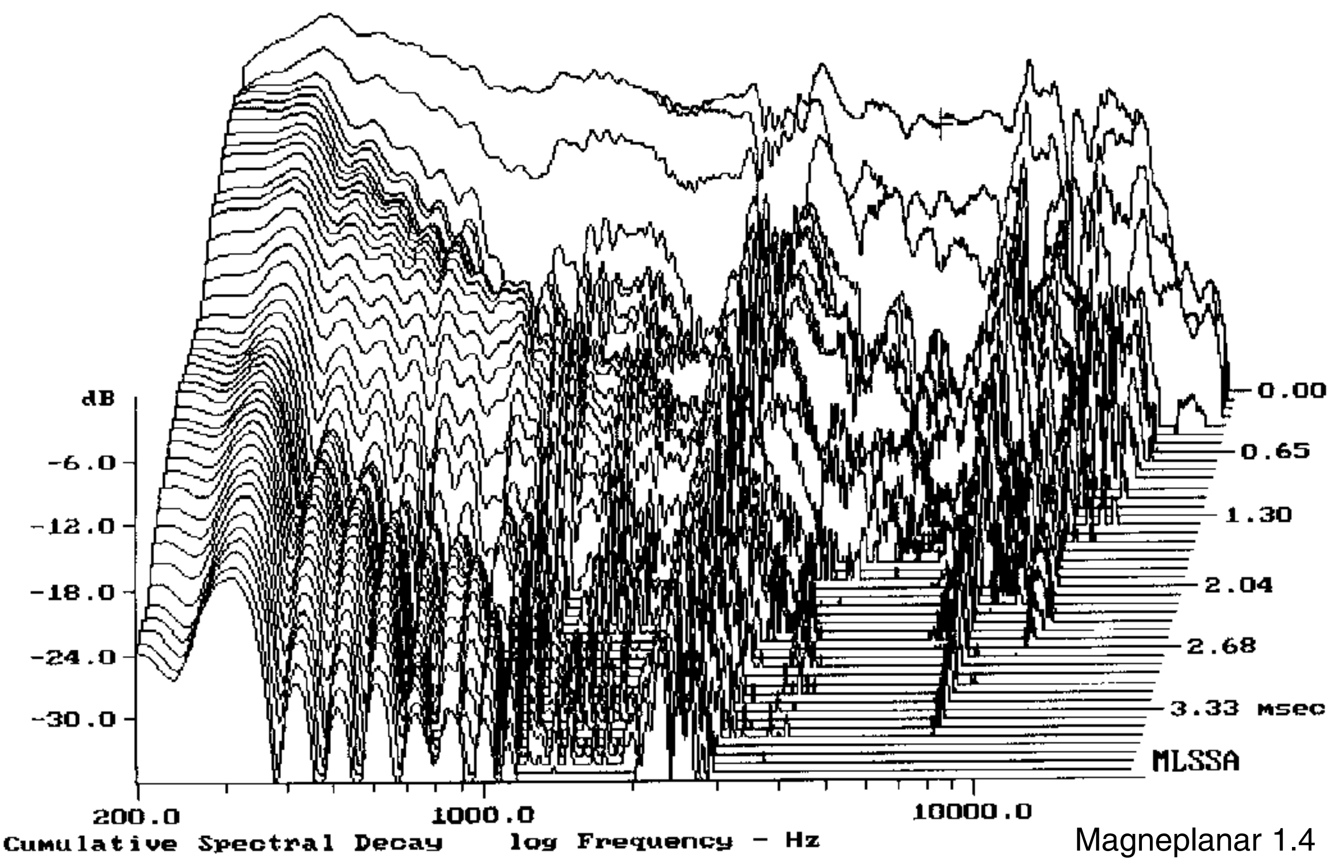
If you casually glance at the graphs for the Kharma Ceramique and the Magneplanar speakers, you may not see much difference. If you refer to the dB scale at the left of the graphs to observe the relative loudness of the delayed energy that each speaker is releasing, you'll see that the Magneplaner is singing alone at a much higher level. After the signal stops, the Martin-Logan speaker continues to sing at high levels as well but it also keeps singing much longer than the others, as indicated by the time graph at the right.

The shape of the energy curves are significant. Random hash that is well down in level will have little audible effect. Steep ridges or valleys that run well into the foreground are likely to be quite audible. The overall amount of energy storage from the speaker is evident in the example graphs.

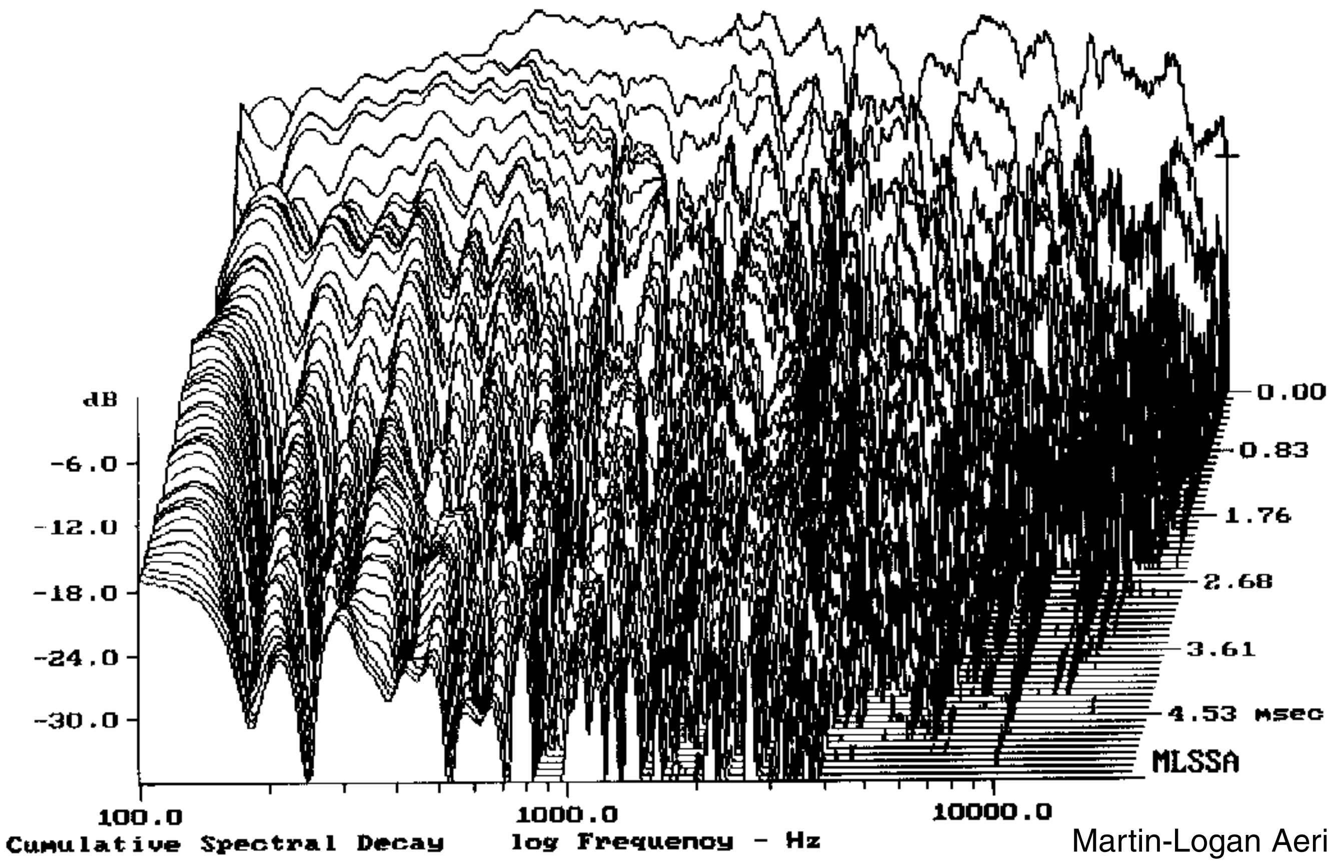
The Kharma Ceramique stores far less energy than the Magneplanar. The amplitude scale for the Kharma speaker goes from -18dB to -42dB and the high frequency ringing is all over after about 2.5ms. (An increase in level of 6dB indicates an output with four times the power. A 10dB increase in level is considered to be twice as loud.) The amplitude scale for the Magneplanar goes from -6dB to -30dB indicating much, much higher levels of delayed output and there are substantial ridges in the midrange and highs that continue to produce relatively high output after 3.5ms. The large ridge centered at 2.5kHz will



Kharma Ceramique



Magneplanar 1.4



Martin-Logan Aerius

provide lots of added "ambience" and the sharp ridge at 8kHz will provide lots of "air" which may partially compensate for the fact that this speaker has virtually no high frequency output at all. The spectral decay of the Martin-Logan speaker looks even worse.

The amplitude scale for the Martin-Logan speaker goes from -6dB to -30dB and the time scale goes to 5ms (a much longer period than the others). This speaker continues to produce strong output across the spectrum for time periods that go off the scale. Inexperienced listeners may perceive these added sounds as "enhanced detail" but delayed energy emanating from the speaker will actually obscure musical detail that follows any moderately loud signal.

The Martin-Logan speaker creates, rather than recreates, a great deal of output across the frequency spectrum.

That's It

In this issue we have discussed speaker measurements and I've shown you some measurements of speakers which are demonstrably inaccurate in one way or another. In the next issue we'll look at some truly accurate speakers and talk about the engineering decisions that made them that way. We'll examine products from three major manufacturers and discuss the compromises chosen by each designer. Yes, even accurate speakers have design compromises, as we'll see in the next issue.