Speaker Measurements 101

How different types of speaker measurements are done—and why some are better than others

By Brent Butterworth

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The advent of cheap audio measurement gear has made it easy for do-it-yourselfers to tweak their systems and even test their own speakers and amps. Unfortunately, it has also spawned a new generation of would-be technicians doing really bad speaker measurements.

You can see this phenomenon on various websites and forums, where the preferred speaker measurement technique is often to plop a measurement microphone 1 meter in front of the speaker, in a typical residential living room, and run a frequency response sweep at 1/24th octave resolution. As we'll see, this technique is like making a really good singer perform an audition next to a guy running a chainsaw—it tells you *something* about the sound, but nowhere near as much as you could tell if you improved the test conditions.

I can sympathize, though. There's not a lot of material about speaker measurement out there. Famed speaker engineer Joe D'Appolito wrote a book about it, but that's a whole book you gotta read. Perhaps the best relatively brief explanation I've seen is in Chapter 14 of the manual for the LinearX LMS audio analyzer manual. Still, there's no simple, accessible primer on how speaker measurements are done.

Until now. This article will take you through the various ways that speaker measurements are commonly done; explain the strengths and weaknesses of each one; and provide some helpful hints about how you can do reasonably good speaker measurements with simple, cheap gear you may already own.

I'll not only explain the different techniques, I'll actually show you the results of each technique on the same speaker. To do this, I measured the same speaker in various ways, using:

1) The best techniques and equipment found in the world's best audio labs
2) The kind of equipment most manufacturers and top audio reviewers own
3) The gear and techniques most enthusiasts employ, and doing it badly
4) The gear and techniques most enthusiasts employ, and doing it well.

I couldn't have done this article without the help of Allan Devantier, manager of acoustic research at Harman International, parent company of JBL, Revel, Infinity, and numerous other pro and consumer audio brands. Allan is the guy in charge of speaker measurement at Harman, and also amuses himself with little side projects like calibrating the sound system at the Staples Center. He suggested using a Revel Performa3 F208 as the test speaker, and I agreed.

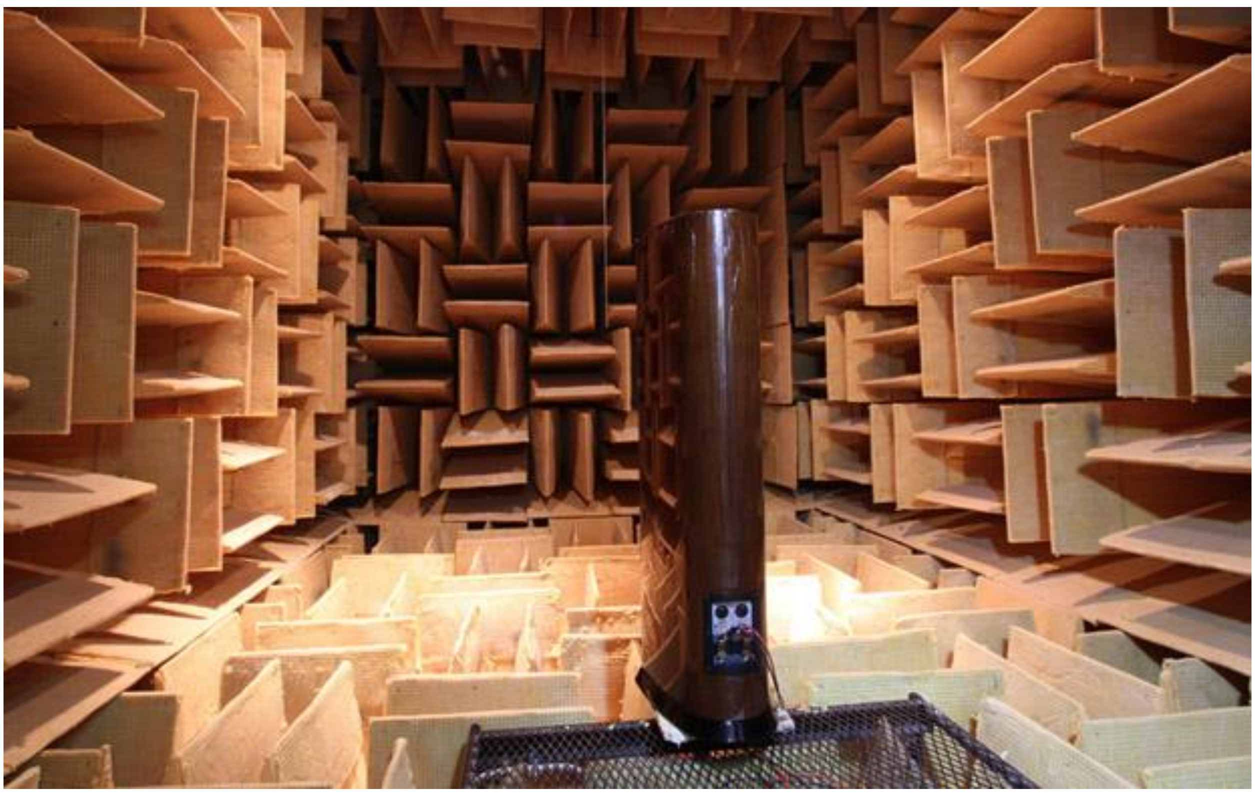
I should also thank two engineers who've been especially generous in aiding my efforts to measure speakers, sharing their techniques while gently pointing out the

errors in mine: Vance Dickason, author of *The Loudspeaker Design Cookbook*, and Paul Barton, founder and chief engineer of PSB Speakers.

One more note: This is a *primer*. For the sake of brevity, I've left out a lot of details. If you want to read something that tells you every last little thing about speaker measurement, try D'Appolito's book.

The Gold Standard: Anechoic Chamber

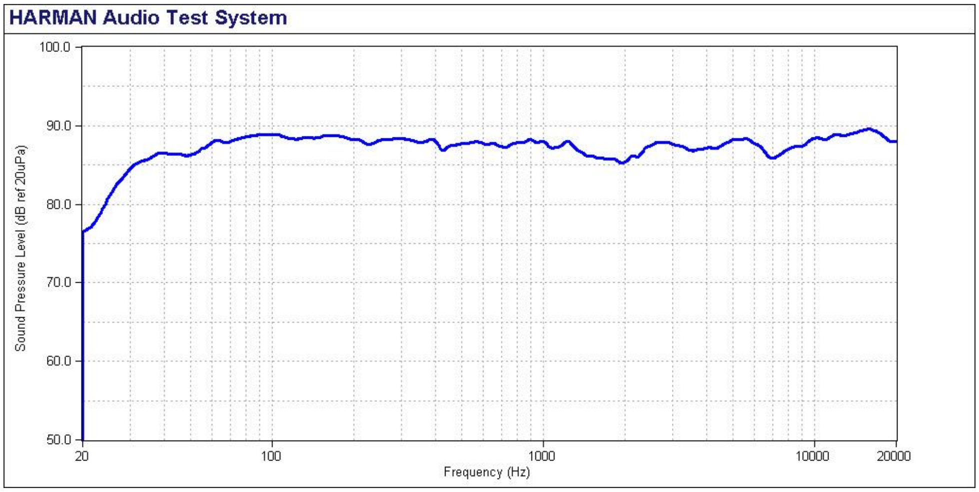
Pros: Easy, accurate, dependable
Cons: Even if you had the space for it, you couldn't afford it



Measuring the F208 in one of Harman International's anechoic chambers

Speaker measurement is complicated because you have to isolate the sound of the speaker from the acoustical effects and environmental noises of the surroundings. For example, if your listening room tends to boost bass at 80 Hz, and that shows up in your speaker measurements, you'll think the speaker has a boost at 80 Hz when it's really the room. Sort of like if you were auditioning that really good singer next to another singer who's out of tune.

The best way to eliminate the acoustical effects of the surroundings, and to eliminate the influence of environmental noise, is by measuring speakers in an anechoic chamber, a space filled with giant fiberglass wedges that absorb all sound. Put the speaker in the chamber, put the microphone in front of it, and it's almost as if you'd hoisted the speaker and mike up a mile high over the Pacific Ocean on a windless day when no ships or planes are passing within 100 miles. It's just speaker and microphone.



On-axis measurement of the Revel F208 taken in Harman International's anechoic chamber, with 1/24th octave smoothing

So why doesn't everyone measure speakers this way? The shell and the wedges for an anechoic chamber cost typically about $500,000. You also need a minimum of about a 30- by 30- by 30-foot space in your building to install them. Once the chamber's installed, you'll still need to invest in measurement gear, including microphones, a computer audio interface of some sort, and a computer with appropriate software.

Even in an anechoic chamber, doing just an on-axis measurement—say, with the mic directly in front of the tweeter—only tells you so much. It's like if all you asked the really good singer you were auditioning to perform was "Mary Had a Little Lamb," because it gives you only a limited perspective of what the product does. Yes, a speaker radiates sound forward, but it also radiates sound in every other direction, too. These off-axis sound waves eventually reach your ears by reflecting off the walls, ceiling, and floor. So if the sound of those off-axis waves doesn't reasonably match the on-axis waves, you've got a problem.

Speaker companies, and most of the pro reviewers who do measurements, measure the response of speakers both on-axis and off-axis. Harman takes it to the extreme by measuring speakers on-axis, then off-axis in 10-degree increments in a 360-degree circle both horizontally and vertically. Often some of these measurements are combined to create an averaged response across a "listening window." Past practice at *Sound & Vision* was to average the results at 0°, ±10°, ±20°, and ±30° horizontally. With the merger of *S&V* and *Home Theater*, this practice will be changing to the standard employed by *Home Theater* and sister publication *Stereophile*: an average of 0°, ±15° horizontally, and ±15° vertically.

The Acceptable Compromise: Quasi-Anechoic

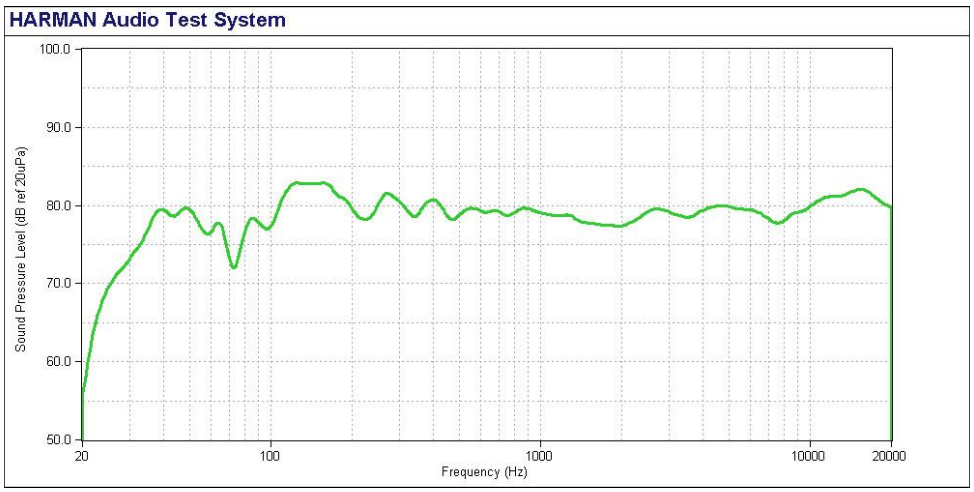
Pros: Reasonably affordable, fairly accurate
Cons: Complicated, only fairly accurate

Measuring the F208 in the author's backyard using quasi-anechoic technique

As I said before, few speaker companies—and no reviewers or hobbyists—have the resources to buy an anechoic chamber. (Although some reviewers, such as the guys at SoundStage!, are lucky enough to be able to rent one on occasion.) Quasi-anechoic technique seeks to simulate an anechoic chamber by electronically removing reflections from nearby objects. As we'll see, it's not perfect, but it's sufficiently accurate and affordable that it's now the preferred technique for reviewers and small speaker companies.

Here's how it works. Imagine a speaker placed on a stand 2 meters high, and a measurement mic placed on a stand 2 meters high and 1 meter from the front of the speaker. At sea level, it takes sound 2.9 milliseconds to travel from the speaker to the mic. The sound that reflects off the floor—the next closest object, assuming a high ceiling or outdoor setting—takes 12.5 ms to get from the speaker to the mic. So if you were to shut the mic off before that reflected sound hits it, you'd have a nice, clean measurement, right? This is called a gated measurement, and it's the core of the quasi-anechoic method.

The problem is, you have only 9.6 ms between the time the sound from the speaker hits the mic and the time the reflected sound hits the mic. To measure an audio signal, you generally need to capture at least one entire plus/minus cycle of it. And if you have only 9.6 ms seconds to work with that means the lowest frequency at which you can grab a full cycle before the mic shuts off is 104 Hz.



Quasi-anechoic on-axis measurement of the F208 taken in the author's backyard, with 1/12th octave smoothing

But it gets worse. To ensure a good measurement, you have to shut off the microphone well before that floor bounce reaches it. So figure we're now up to 120 Hz. But the lowest frequency you can catch before the gate closes is *also* the resolution of your measurement for the first octave. The resolution doubles with each octave, so you're up to 1/2-octave resolution between 240 and 480 Hz, 1/4-octave resolution between 480 and 960 Hz, etc.

The result is what I call "the squiggles": wavy lines in your frequency response measurements below about 1 kHz that you know from other measurement techniques (which we'll get to) shouldn't be there.

There's another problem, too: How, then, do you measure the response of a speaker at low frequencies? This part's not actually so hard. There are two pretty good techniques you can use:

1) Place the mic close to the woofer, which makes the woofer so loud relative to the room acoustics modes that the effects of room acoustics don't appear in the measurement. Repeat this with the other woofers, the port(s), and the passive radiator(s), then sum those measurements together to get the total bass response.

2) Run a ground plane measurement by placing the speaker on the ground and the microphone at a distance of 1 or 2 meters (depending on the size of the speaker), then using an EQ correction to remove the effects of the room acoustics (although if you do this in a large open space you don't have to worry about the EQ correction).

I use both of these techniques, the choice depending on the size and configuration of the speaker. Close-miking produces a smoother-looking response curve, but it also tends to produce a bump in the bass response, typically around 100 Hz, that's not actually there.

In both cases, the bass response is then spliced to the quasi-anechoic response at a frequency between about 150 and 300 Hz. There's still one problem, though: The level of the bass response curve will be higher because of the closer mic position, or because the ground-plane technique boosts output by +6 dB. There are formulas you can apply to get the "proper" scaling, but I and most of the technicians I know just eyeball it, lowering the bass curve until a big part of it overlaps with the quasi-anechoic curve, then making the splice.

All of the quasi-anechoic measurement packages I've used (LinearX LMS, Audiomatica Clio 10 FW, PureBits Sample Champion, and Listen Inc. SoundCheck) employ some combination of averaging and filtering (i.e., filtering out frequencies other than the one being tested at that moment) to reduce or eliminate the effects of environmental noise. This usually works very well. Once my neighbor's gardener started up a leaf blower while I was running sweeps with LMS in my backyard. To see how well LMS could filter out the noise, I ran a speaker measurement while he was working and another after he stopped. There was no significant difference between the two curves.

When you're using quasi-anechoic technique, it's easy to put the speaker on a turntable, just as they do in anechoic chambers, to measure the response at various off-axis angles. These can then be averaged and compared with on-axis response, as I've done in many of my measurements for *S&V*.

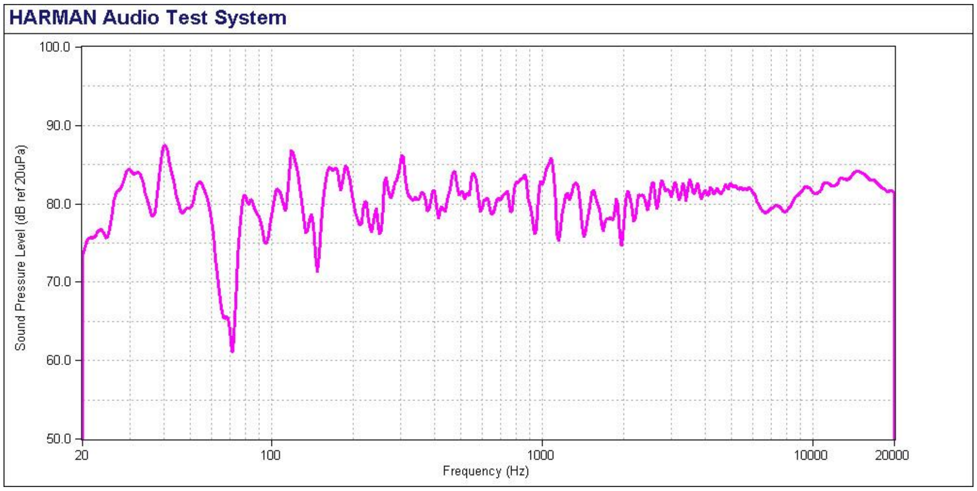
The Worst Way: In-Room, Unaveraged, Unsmoothed

Pros: Easy, cheap
Cons: Marginally useful

Measuring the Revel F208 at a distance of 1 meter in the author's listening room. Don't do this.

The easiest possible way to measure a speaker—or really, approximate measuring a speaker—is to do as I described in the introduction: put a microphone in front of it, run a frequency-response sweep or pink noise through it, and measure the result with a real-time spectrum analyzer, such as TrueRTA. Honestly, this is what I did the first time I tried to measure the response of my audio system more than 20 years ago, using an old-school AudioControl dedicated analyzer with an LED display. In most cases, this measurement is usually done at the maximum resolution of the analyzer, typically 1/24-octave, and it's usually done only on-axis.

The problem with this technique is that it doesn't accomplish the goals of good speaker measurement. It doesn't isolate the speaker's response from the effects of room acoustics, or from reflections of sound off nearby objects. It doesn't let you separately analyze the on- and off-axis response, because even though your measurement is primarily on-axis, you're getting some of the off-axis response, too, after it's bounced around the room a bit.



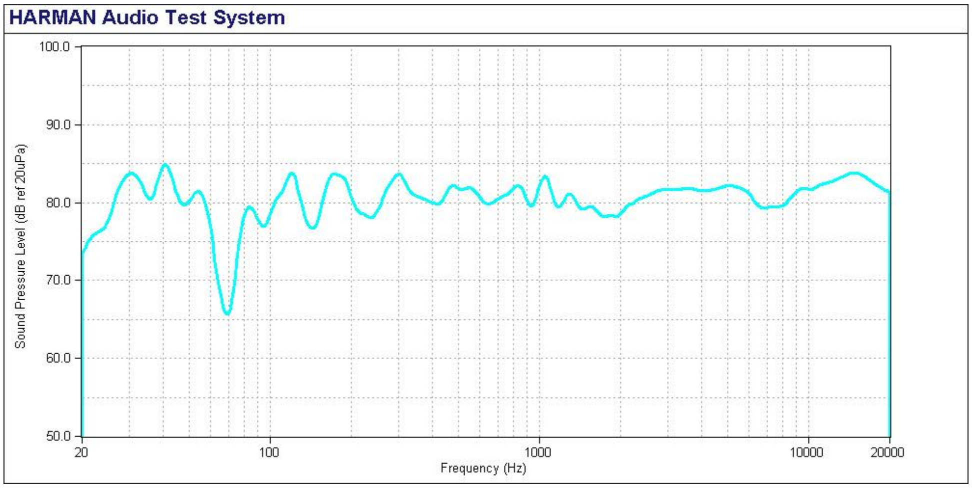
In-room, unaveraged measurement taken at 1 meter in the author's listening room, with 1/24th octave smoothing

Does this mean that in-room measurements are of no use at all? Absolutely not. Let's consider some ways to improve them.

First, measuring in-room at 1/24-octave resolution doesn't really tell you much about how a speaker sounds. It just gives you a frequency response curve with a lot of hash that obscures what the speaker's actually doing. It's sort of like trying to judge a model's looks solely by viewing her skin through a magnifying glass—you see the details but you miss the big picture.

If you compare a 1/24-octave in-room measurement with a 1/24-octave anechoic or quasi-anechoic measurement, you'll see that the latter measurements don't have all that hash. So where's the hash coming from? It might be a performance issue with the speaker. More likely, though, it's reflections from objects in the room, which have nothing to do with the speaker's performance. As I said in the intro, it's like auditioning a singer while a chainsaw's running in the same room. The worst part is, even if the peaks and dips are caused by a flaw in the speaker, they are often inaudible even though they may look really scary.

If you smooth the curve by cutting the resolution down to 1/12-octave or even 1/6-octave, you start to get a picture of the speaker's tonal balance. Get rid of all that hash and the major characteristics of the speaker's response and tonal balance—the things you *can* hear—become apparent.



In-room, unaveraged measurement taken at 1 meter in the author's listening room, with 1/6th octave smoothing

You still have a couple of problems, though. Worst is that the room modes—i.e., the natural bass resonances of your room, which occur below the Schroeder frequency—are showing up in your measurement. This is also easy to fix.

If you play a bass tone, or even a tune with a consistent bass line, and walk around the room, you'll hear the sound of the bass change quite a bit. That's because the bass waves reinforce each other in certain places and cancel in other places. You can take advantage of this by simply taking measurements in several places in the room, then averaging those measurements. Most of those big peaks and dips you've been measuring in the bass will suddenly disappear.

Averaging measurements from several positions has another benefit: It gives you a good mix of on-axis and off-axis sound, approximating the listening window curve I discussed previously.

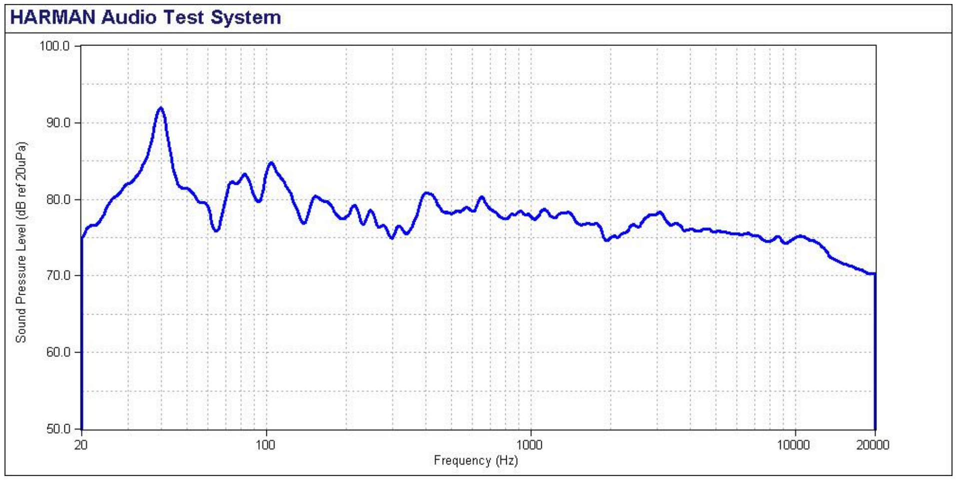
A Pretty Good Way: In-Room, Averaged, Smoothed

Pros: Easy, cheap, captures what a speaker does in a real room
Cons: Doesn’t allow much technical analysis of a speaker

Measuring the F208 in the author’s listening room using averaged in-room technique

About 15 years ago, an off-hand comment from Scott Bagby of Paradigm changed my life—or at least the part of my life where I’m dealing with speakers. Hearing of the difficulties I’d had in scaling and summing bass responses, then finding the right place to splice them to quasi-anechoic response curves, he said, “Doing an average of five or six in-room measurements from different mic positions can tell you a lot about a speaker.”

Boy, was he right! Since then, if I ever feel my bass measurements might be off, I’ll do an averaged in-room measurement to confirm. *Stereophile* does an averaged in-room measurement along with quasi-anechoic speaker measurements, to give a “real world” portrayal of how a speaker performs in an actual room. (Of course, to do this, you need real-time analyzer software that can perform magnitude averages. TrueRTA, an affordable and popular PC app, can do this easily.)



Averaged in-room measurement of F208 in the author's listening room, combining the results of 12 different microphone positions

But where do you put the microphone? What different positions should you use? I tend to put the microphone at my seated ear height, take one measurement with the mic where my head would normally be, then move the mic about 2 feet to either side then 2 feet forward and back, staggering all the mic positions slightly so no two are ever in a line parallel to the room's walls. *Stereophile*'s John Atkinson places the microphone in each corner of a vertical rectangular grid 36 inches wide by 18 inches high, centered on the positions of his ears. I've seen manufacturers and installers place a microphone in each of a room's seating positions, with the mics all at or near seated ear height.

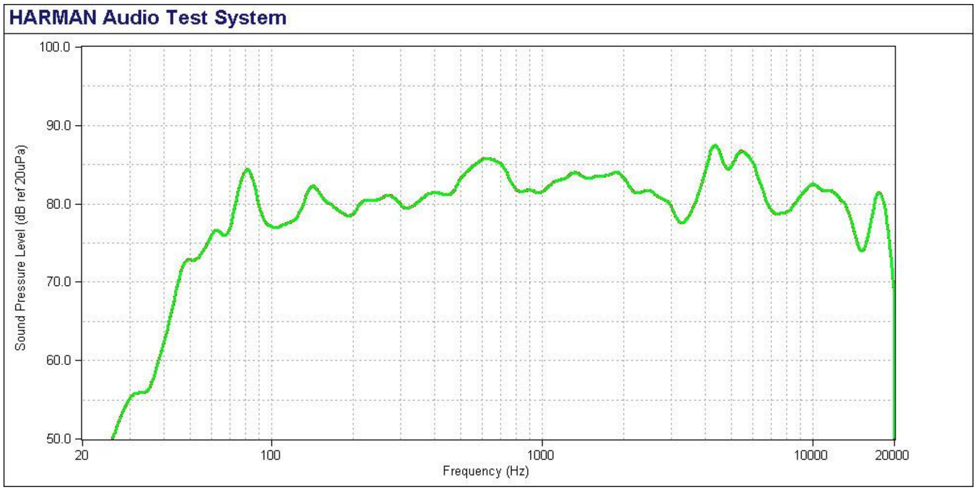
I'd hoped Devantier would give me some inviolable golden rule of in-room measurement mic positions, but when I asked which technique is best, he simply said, "We don't know." But he quickly added, "It doesn't make that much difference, as long as you're moving the mic a couple of feet in each direction."

Once you have your five or six or eight response curves taken at different mic positions, you can average them all together to get your speaker's averaged room response. Then you can apply smoothing. Start with 1/12-octave, and if that's too hashy, try 1/6-octave.

According to Devantier, there's no firm standard yet for what an ideal averaged in-room response looks like. But the frequency response plot should look fairly smooth, and should slow a mild downward tilt in the tonal balance, with the peak bass output perhaps +6 dB greater than the response at 20 kHz.

Still, though, I wasn't quite satisfied. The in-room averaged technique had given us some meaningful results when we measured one of the industry's most technically perfect speakers. What, though, would happen if we measured a less-than-perfect speaker? Would its flaw show up in the measurement?

To test this, Devantier and I measured a speaker I made using an Audio Nirvana 6.5-inch full-range driver with a whizzer cone—a nice driver, but one built using a primitive, circa-1940s design that some audiophiles claim to prefer. You can see the result in the graph below. Judge for yourself.



In-room, averaged response of the built with 6.5-inch Audio Nirvana full-range driver

While you probably can’t use averaged in-room measurements to do in-depth analysis of a speaker, such as the effects of its crossover, it will, as Bagby suggested, give you a quick and accurate picture of what you’ll actually hear from a certain speaker in your listening room. And I can promise you, if you ever tell a speaker engineer that you’re measuring your system this way, he’ll approve.