Equalization Revisited:

By John Murray

Mixing Technicians and Creative Equalization

There are two separate and distinct disciplines in live sound reinforcement systems that employ equalization. The first is the mixing technician. This discipline is done using the tools on the mixing console, like level balancing and source equalization, along with all the tricks and tools available in your typical effects rack, and perhaps most importantly, source microphone selection and placement. It takes years to really master this array of devices and those that do are truly artists in my book. Same thing goes for a monitor mixer. Equalization, as a creative effect, is to paint the mood and emotional expression of the music. It is also used to correct response problems in source microphones. It is identical to what is done in the recording studio, and *is never done downstream of the mixing console's output*. This article will not address this type of equalization.

System Technicians and System Equalization

The second discipline addresses all the adjustments made downstream of the console's output. It is not artistic at all and is commonly known as system engineering. It is highly technical, involves impedance, phase, frequency response, signal delay, compression ratios, gain structure, FFT's, etc. and is very mathematic in nature. It is really a technician's position, not an engineer's (let's keep those egos in check, shall we?). It is where I live in this industry and have spent 26 years trying to understand all that is involved with this discipline.

Once a signal leaves the console, all the devices employed downstream have one function: to route and condition the signal for reproduction by a speaker That's it. The idea is to *make that acoustic signal one hears in front of a speaker as close as possible to the one that leaves the console*. Why does SMAART have a transfer function mode that compares the electrical source signal to the acoustical signal and enables you to remove the difference? Why do studio monitors strive to have an absolutely uncolored sound that enables you to hear exactly what is being recorded?

This fact is what makes sound-system equalization a cut-and-dried, strictly objective, by-the-numbers technical procedure. It is not subject to interpretation, feel, or mood. If the correct data is collected, the correct equalization can be done so that the signal is as close to the original as is practical. The question is, then, how is this correct data collected?

EQ: Ear vs. Measurement?

Which is the best way to equalize a sound system, by ear or measurement? The short answer is both. Each method compliments the other. The ultimate qualification for sound quality is the ear. If it doesn't sound right, nothing else matters. I once spend 2+ hours tuning a church sound system using the methods I have used on over a hundred speaker systems over the years and the resulting sound sucked. Rather than trying to convince my ears that the very good curve on the laptop screen sounded just wonderful, I had to trust my ears that something was wrong and admit that to the customer standing there. It turned out that my calibrated mic was damaged and once the process was repeated with an undamaged mic, the sound quality matched the curve. The ear is the final judge, and no matter how enamoring the technology, common sense (one of the five!) must prevail.

However, due to things like illness, drug effects, poor acoustic memory, which we all have, and hearing deterioration due to age, tuning a system purely by ear will not produce consistent sound quality nor the absolute best a system can do. Clearly, a system tuned by measurement will find problems that the human ear just isn't very good at detecting, but are ones that make the difference between an OK sound and a spectacular sound, *and will do it consistently*. I would challenge anyone to find the one out-of-polarity transducer in a large system by ear. You might be able to tell something is wrong at a particular listening position, but you just can't tell exactly what. Measurement pinpoints that type of problem exactly.

What the Ear Does Better

There are things, too, that the ear is better at than measurement systems. One example is balancing levels between separately amplified multi-way speakers. Level matching the two jagged response curves of a woofer and HF section on a computer screen is much more difficult to get right than just doing by ear. This is particularly true when changing just 1.0 dB of relative level can completely change the character of the speaker system.

Operator Error

There are many systems that have been TEF'ed , SIM'ed, SMAART'ed, etc. that plain just don't sound good. So what is the issue? Why have some people just given up and think that measurement systems just don't work? I can tell you that it is not problems with the measurement systems. They are improving all the time. The largest problem, by far, is operator error. These operator errors fall into three general categories: 1) the flat RTA response misnomer, 2) improper measurement-mic placement, and 3) attempted equalization of multiple-source or multiple-reflection-contaminated, FFT measurements.

It has been my experience that 95% of all sound systems are equalized improperly due to these three errors. It is my belief that this is why some road dog types thoroughly mistrust measurement geeks. The large majority of them simply do not use the measurement system correctly. In the case of the newer systems like TEF, SMAART, and SIM, they don't put the measurement mic in a correct location for what they are measuring, and they try to equalize multiple transducers at short wavelengths.

The Flat RTA Misnomer

An RTA (Real-Time Analyzer) is a 2-dimensional measurement system, which displays energy in dB SPL or volts vs. frequency in hertz. TEF, SMAART, SIM, etc. are all 3-dimensional (3-D) measurement systems that display energy vs. frequency vs. time. Therefore an RTA, unlike FFT (Fast Fourier Transform)-based 3-D measurement systems, is time blind and lumps all energy occurring within a fraction of a second together. A fraction of a second is an eternity to a 3-D measurement system.

If you measure an electrical voltage signal, like pink noise at the output of an analog mixing console, the mixer's electronics have very little propagation delay. Electrical signals on an RTA display will very closely match what 3-D FFT's display. This is because electronics do not *time-smear* the original signal. Therefore, if an electrical pink-noise signal is flat on an RTA, it will also be flat on a 3-D measurement system as well.

However, once an electrical signal is converted by a speaker to an acoustical one and reflected around a room, the time smear is substantial. All the energy is not present at the same point in space at a single point in time, nor is it all dispersed from the speaker uniformly with respect to frequency. The direct sound signal that travels straight from a speaker to a measurement mic will be the shortest path and travel time between the two. Energy that first reflects off a side wall, then a back wall, then the floor, then to the mic, will take many milliseconds more. This later energy appears to arrive simultaneously with the direct sound on an RTA display and will be summed with it. However, this reflected energy can be ignored by the display of a 3-D measurement system, and it is this characteristic that makes it a superior measurement system.

If you equalize a speaker to be flat on an RTA display with the measurement mic in the middle of the listening area, you will be unpleasantly surprised by the resulting bad sound quality. This is not a similar measurement to the electrical one by any means.

The Room Curve

Probably the first person to recognize this difference was Dr. Boner, the godfather of audio consultants who was one the first to practice equalization. He developed what has been called the *ideal-room curve* or *preferred-listening curve*. It is largely the acoustical *power response* of the speaker system, as modified by air and surface absorption within the room (See **Figure 1**).

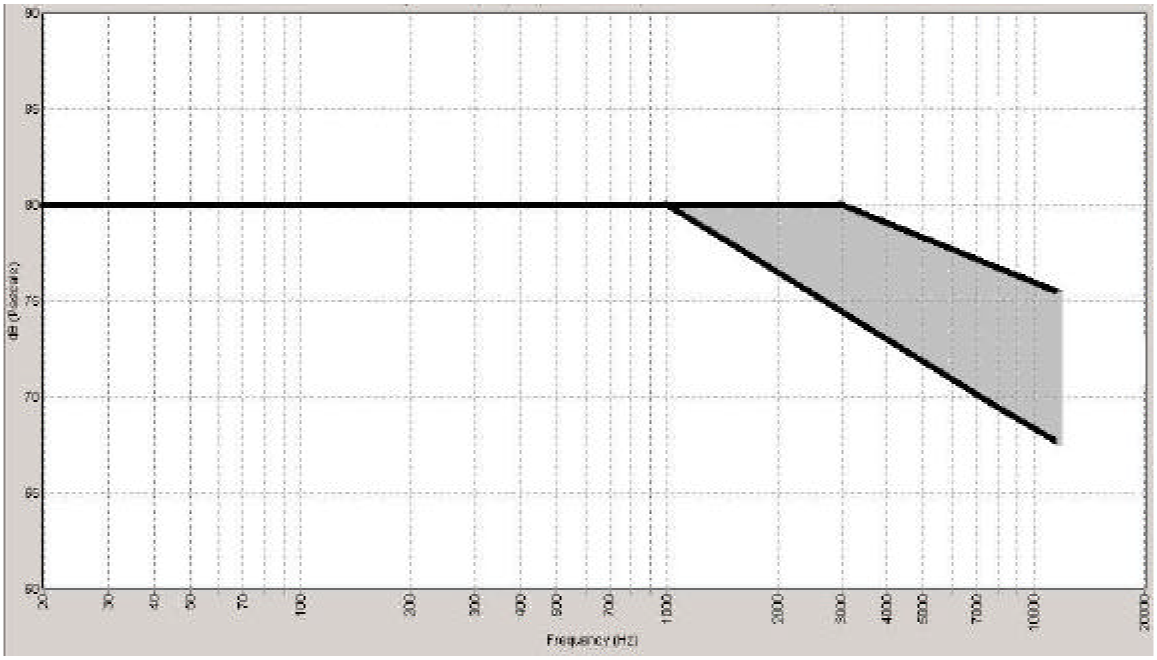

Figure 1: The *ideal-room curve*, or *preferred-listening curve* with its range of high-frequency variation in gray. Note that the lower limit is the original standard for cinema sound systems with beaming, radial-derivative, high frequency horns. The upper limit fits better for more-recent, constant coverage horns.

Power Response

What is the power response of a speaker system? Other than being one of the most over-used and least-understood terms used today in audio, it is the sum total acoustic power that a speaker produces. For example, let's measure a two-way speaker system in a large room, with the measurement mic in front of the speaker in the middle of the listening area, in the reverberant field beyond critical distance, where the direct sound is lower in level than the reverberant field energy. Most all of the high-frequency energy is aimed in the general direction of the mic due to the directional effect of a horn on the HF driver. However, most of the low-frequency energy is not aimed at the mic because the LF driver is omni-directional for most of its passband.

The Direct Sound is Hidden in the Room Curve of an RTA

Therefore, if a flat anechoic or direct sound is desired, much more energy must be generated into the room by the LF driver to equal the SPL of the HF driver at the mic's position. On an RTA display, this will look like the LF is a big haystack and the HF gradually rolls off toward the higher frequencies where its horn exhibits better dispersion control. Obviously, this is Boner's curve. Where the exact hinge-point of the HF roll-off begins and just how steep the roll-off is depends on the dispersion of the HF horn, the number of devices, and whether the LF section has any directional control or not.

In a movie theater, where the size and absorption characteristics of the room, number, location, and specs of the speakers are all fixed, a tightly defined curve can be used. For us sound reinforcement folk, where every room and speaker system is different, the modified power-response curve that produces a flat direct response can vary a lot.

In the days before 3-D measurement systems, one had to vary the hinge-point and roll-off characteristics of the curve until everything sounded right. This took a lot of time to get a satisfactory result. It had to be done with each individual system until manufactured one or two-box systems (mid-high packs and subs usually) came along.

With a manufactured system like an EAW KF-850, you got a feel for what variation of the curve worked for that system. You turned on all the speakers, put the RTA in the reverberant field, and EQ'ed until you got that curve that seemed to work best.

In the Direct Field

If the mic was too close, and in the direct field rather than the reverberant, the curve changed with each move of the mic. Often in smaller clubs, you couldn't get it far enough to be in the reverberant field. The measurement-mic position caused the RTA to partially reveal cancellations due to multiple-source or refection cancellations. This condition necessitated winging the EQ curve by ear. Thus the "tune-by-ear" approach to system EQ was born. Once 3-D measurement systems came along, a completely different approach was needed. Few system technicians were trained on this approach.

And that brings us to the second and third operator-error categories.

Time-Oriented Misalignments

This ten-dollar phrase describes time-oriented events that cannot be equalized. Examples of this are other delayed sources, like reflections or more distant speakers, which are delayed enough in time to cancel the direct-sound energy from a speaker at particular frequencies (see **Figure 2**). Another is the notch at the crossover frequency of a speaker system when the drives are not time-synchronized (see **Figure 3**). Neither of these frequency-response problems can be remedied by equalization. These are time-oriented mis-alignment events and cannot be fixed with EQ. This also applies for reverberation or echoes. Even a change in the reverberant nature of a room, due to a change in its acoustical absorption characteristics, is not an equalizable situation.

"Room EQ" Changes

Yes, I can already hear the protests to that last statement, "But I've had to change the "room EQ" numerous times when it was EQ'ed empty early in the day, and didn't sound right when the room filled with people and the room was no longer as reverberant." Yes, the EQ had to be changed, but it was not due to reduced reverberation in the room. What had to be accommodated was the effect of temperature and humidity changes on the direct sound from the speakers, not the reverberation. These effects on the direct sound from the speakers are not time-oriented, and as a result can be equalized. Why else would a BSS OMNI-Drive have a meteorology probe? To measure reverb time? No, to measure changes in temperature and humidity, and to change the EQ curve accordingly.

New Measurement Systems

So now you've gotten your new 3-D, FFT-based measurement system and you're going to tune your first system. The first thing you do is place the calibrated mic on a mic stand in the middle of the listening area, turn on all the speakers in the system, and turn up your test signal, just like you did when using your old RTA. Right? Absolutely wrong!

Measurement Microphone Position

First, putting the mic on a common mic stand contaminates the FFT-based measurement unnecessarily. That hard reflection off the floor is a time-oriented event, a delayed source whose effect cannot be equalized. For SMAART and SIMM measurements, the mic should be PZM'ed on the floor. This removes the comb-filtering effect of the reflection from the measurement. If you use windowing, you may also put the mic on a light stand to get it far enough away from the floor so that the reflection is delayed enough to be "windowed" out of the measurement time limits. This will also limit the low-frequency response of the measurement, but room modes render low-frequency measurements within a room and far from the speaker useless anyway. More on this later.

Reflective Surfaces Near the Speakers

Do be aware that any hard, reflective surface near a speaker, rather than the measurement mic, will also cause the same problem. Either move and aim the speaker to avoid creating these reflections, or treat the surface with absorption to prevent them. Otherwise, a comb filter (see **Figure 2**) will color the source's tonal quality for most listening positions for that speaker.

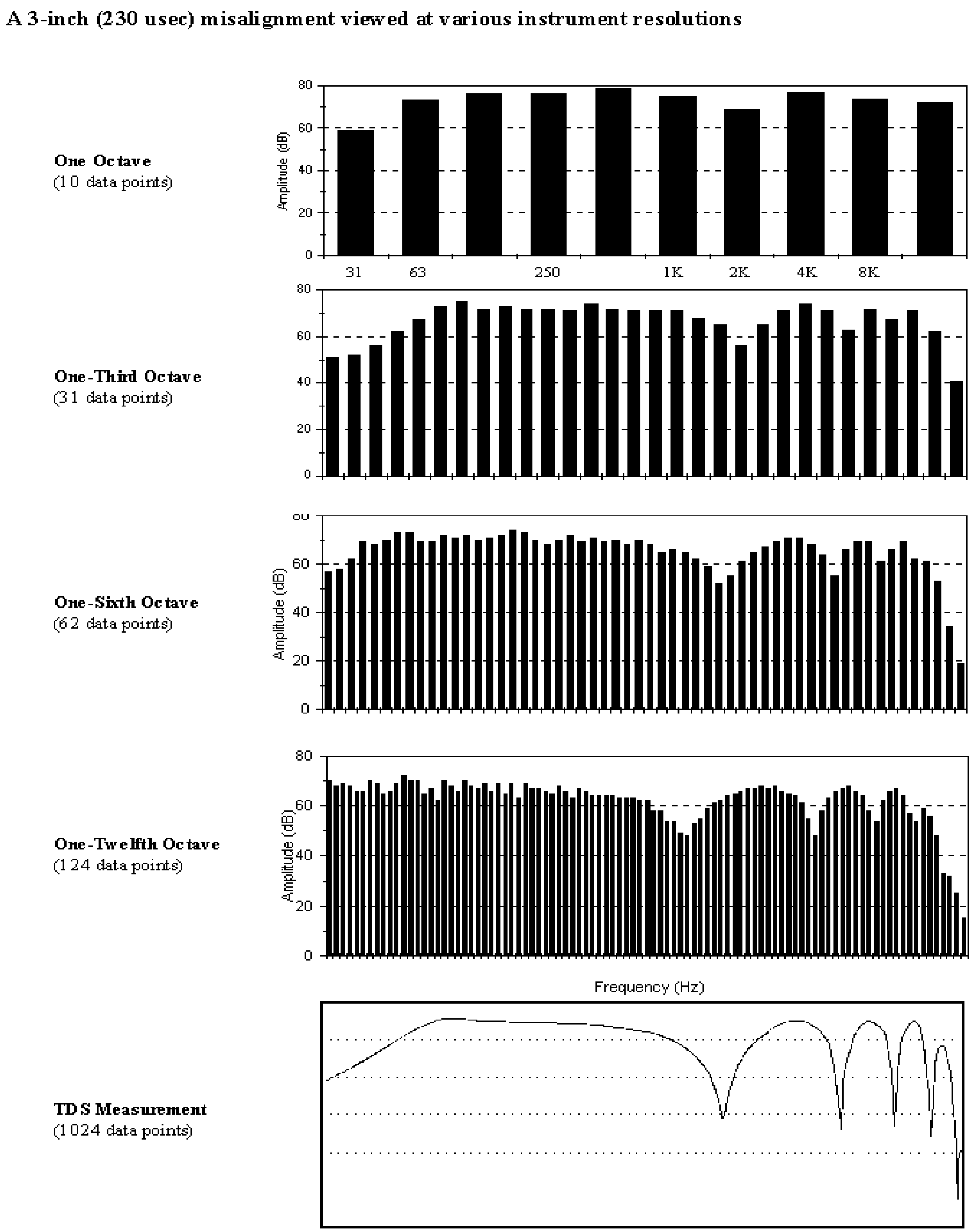

Figure 2: Various measurement-resolutions of a comb-filter caused by a 3-inch signal delay. Note that even the 1/3-octave resolution cannot clearly show that the frequency-response problem is time-oriented, and therefore cannot be equalized. (courtesy of Pat Brown, Synergetic Audio Concepts)

Multiple-Source Contamination

Because 3-D measurements are time-sensitive, other speakers cause comb-filter contamination just like reflections, and must be eliminated from the measurement (see **Figure 2** again!). That's right, you need to be equalizing for a flat direct response on axis to just *one* speaker per crossover-bandpass section. Where long wavelengths combine, like subwoofer and the low-end of low-mid speaker frequencies, further consideration is needed, which we'll discuss later. But for mid-highs, this is a hard and fast rule. Any other frequency-response problems, such as lack of HF level at listening positions off-axis or between cabinets are issues of inadequate coverage, not equalization.

Since we are equalizing speakers and not rooms, particularly at short wavelengths, separate EQ is needed for each type of speaker. One equalizer cannot be tuned correctly if it is feeding multiple types of speakers. You need a dedicated EQ and amplifier signal path for each type of speaker.

If a cabinet has double 12" or 15" woofers, when you equalize the measurement mic must be positioned exactly between the two drivers so that the path-length from each are close enough to not have any cancellations at the frequencies the drivers are reproducing. If a cabinet has two tweeters reproducing the same frequencies, the mic cannot be placed accurately enough to prevent cancellations at very short wavelengths, so proper EQ'ing is not possible. The author recommends permanently disconnecting one tweeter because all listening positions will have high-frequency-response problems with this poorly designed speaker.

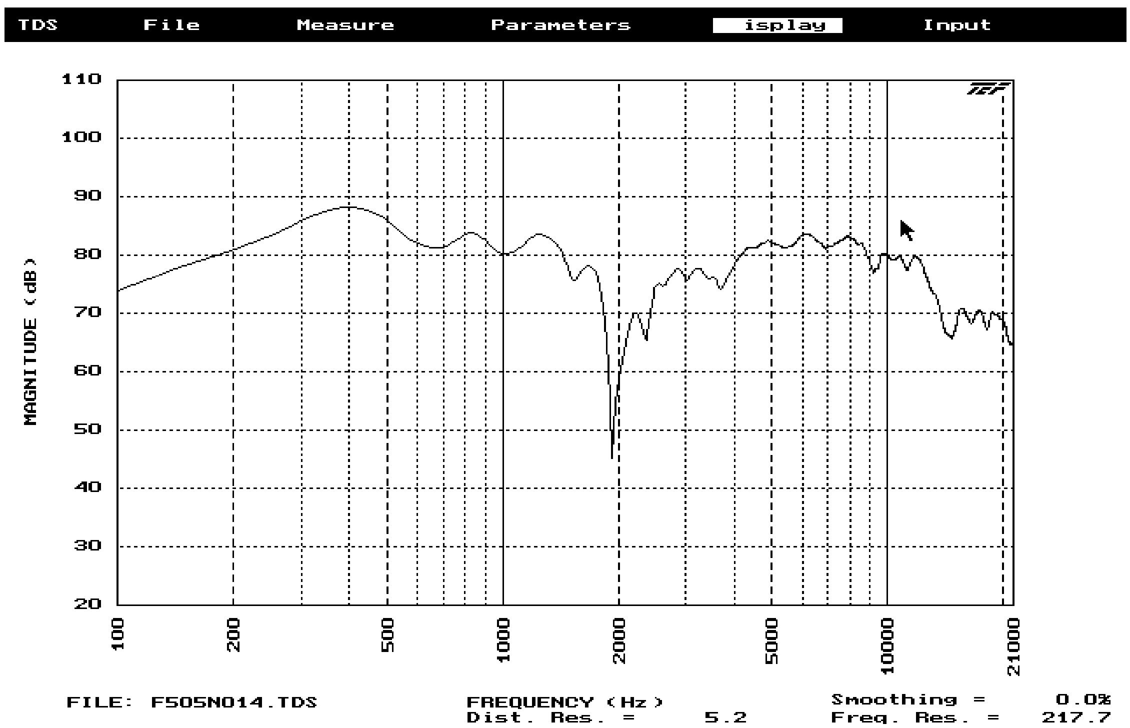

Figure 3: Non-synchronized LF and HF drivers cause the notch at 2000 Hz. Since this problem is due to the later arrival of the signal from the system woofer, it is a time-oriented event, and is also not equalizable. Viewed by an RTA in the reverberant field, this notch would not be revealed.

Diffraction and Co-Axial Devices

Another concern is diffractional effects, particularly in the case of co-axial drivers. Due to the symmetric nature of most co-axial designs, diffractional effects can be emphasized. That is why Frazier takes great pains to add absorption to minimize diffractional effects for their co-ax systems. The author recommends averaging several measurements across the coverage pattern of co-axial drivers and equalizing the average response.

Once the mid-highs have been EQ'ed using one driver per passband, and without any strong reflections contaminating the measurements, then we are ready to tackle the lows.

Room Modes

Room modes are summations and cancellations of long wavelengths whose dimensions are similar to the dimensions of the room. They cause additions that can be 12 dB or more where reflections sum and cancellations 20 to 30 dB deep where they are out-of-phase. Rooms with regular dimensions that are multiples of each other, like a 10-ft. ceiling, 20 ft. across, and 80 ft. deep, or rooms containing circular arcs have the strongest room mode effects.

For example, generate a 100 Hz tone into a room with one speaker. Walk the room and listen for hot spots and null points. They will be 1/2 wavelength apart (1130 / 100 =11.3 / 2 = 5.65 ft. spacing). The way to determine which frequencies are affected by room modes is most easily done using the simpler of two available equations for this.

$$\mathbf{R = (3 \times SS) / RSM}$$

R = the upper frequency limit of room modes
SS = the Speed of Sound
RSM = the Room's Smallest Dimension (usually the ceiling height)

So, for a room with a 24-ft. ceiling height, like a high-school gym, all the frequencies below 141 Hz cannot be equalized with the measurement mic placed anywhere in the room. Clearly when you have easily 30 dB variations in level around the room at a given bass frequency, you cannot equalize those frequencies in the room because the level is wildly position-dependent. What do we do?

Ground-Plane or Extreme-Near-Field Measurements

There are two choices: 1) EQ the bass tones with the speaker system outdoors, like in a parking lot with no reflective surfaces nearby, and once again PZM the mic element against the blacktop for a ground plane measurement. 2) Use D.B. Keele's extreme-near-field measurement technique described in a 1974 AES paper. This method places the measurement mic capsule about 1/8" off the woofer's dust cap. For bass-reflex cabinets, the lowest 1/2 octave of response needs to be tuned with the mic in the port. I have been using this technique for years now with excellent results.

Now that we have equalized both the mid-highs and lows, *which is a process completely independent of the room the speakers are in,* we are now ready to consider the acoustical environment's effect on equalization.

Mutual Coupling and Fractional-Space Loading

The only portion of a speaker's response that the room has anything to do with is where fractional-space loading occurs. This is in the long wavelengths where in-phase summation with nearby surfaces is an equalizable situation. When a speaker is against a floor, wall, or corner, the normally omni-directional bass energy sums in one direction and adds 6 dB, not 3 dB, for each doubling of surfaces (sources) at the longest wavelengths. This bass build-up is the one instance where "room EQ" actually exists. Similarly, multiple low-frequency drivers act just like surfaces and add 6 dB per doubling of sources and this also must be accommodated in your final EQ curve.

However, because of room modes, you cannot place the mic out in the room where it will receive energy from all the surfaces and/or LF drivers, because the bass response will still be wildly position dependent. The solution is that the ear is still the best way to adjust the bass response. Listen in a few different positions to variety of bass sources and you'll quickly be able to determine a good LF balance. I generally just adjust the drive to the system subs (sub-crossover section or power-amp attenuators) in 3 or 4-way systems as a gentle, relatively broadband approach to taming low-frequency build-up that changes very gradually with frequency.

Air Absorption

The air has a very definite absorption characteristic for short HF wavelengths. If the throw of your system is very long (hundreds of feet), you may want to consider a HF EQ boost to counter those effects. Keep in mind that this must be compromised between the closest and farthest listeners. Don't make it too bright for those up close. Also, most people that choose a position far from the speakers naturally expect less HF content, so a flat response to 15 kHz or so might be disconcerting at 100 feet or more. Lastly, too much HF boost can fry you HF drivers. Be careful.

Conclusion

Once you have equalized you system using the method described here, you will find it works quite well and can be applied to any sound system. The sound you will get will be natural and very consistent from system to system. It will also be the best sound you can get, regardless of the room acoustics. The sole exceptions to this are “boomy” rooms with VERY strong room modes. They requirethat the system does not excite the room modes present in the listening area by removing them from the sound system’s response using notch filters.

I have been using this method to EQ systems for many years and it always surprises people that I can tune a system within 95% of its final result before listening to any program source at all. Now, you can too.