What's The Measurement?

Understanding and properly using RTA & FFT

By John Murray

What's the best way to equalize a sound system – by ear or by 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 spent two-plus hours tuning a church sound system using the same methods I've employed on more than 100 systems, and the resulting sound was terrible.

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. It turned out that my calibrated microphone was damaged. Once the process was repeated with an undamaged mic, the sound quality matched the curve.

The ear is the final judge. No matter how enamoring the technology, common sense must prevail!

However, due to things like illness, drug effects, fatigue, poor acoustic memory (from which we all suffer) 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. Tuning by measurement will uncover problems that the human ear just isn't very good at detecting, but they are problems that make the difference between sound that's "O.K." as opposed to spectacular (and consistent).

I 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.

Don't get me wrong – the ear is ideal for certain things. For example, balancing levels between separately amplified multi-way loudspeakers. Level matching two jagged response curves of a woofer and high-frequency section on a computer screen is much more difficult to get right than by ear. This is particularly true when changing just 1 dB of relative level can completely change the character of the speaker system.

And there are many systems that have been "TEF'ed, SIM'ed and SMAART'ed" that plain just don't sound good. Could this be a big reason some sound folks think that measurement systems just don't work? I think so.

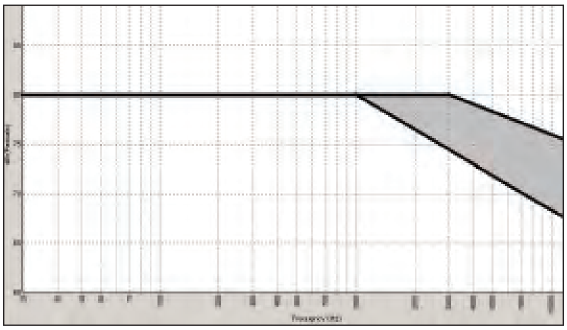

Figure 1: The ideal-room curve, or preferred-listening curve with its range of high-frequency variation in gray. Note that the lower limit is the original standard for cinema sound systems with beaming, radial-derivative, high frequency horns. The upper limit fits better for more recent, constant coverage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

But the issue lies not with measurement systems, which are improving steadily. No, the largest problem, by far, is operator error. Rather than continue the debate, let's get busy with addressing the errors that commonly plague us as we tune by measurement.

THE BIG THREE

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's been my experience that 95 percent of all sound systems are equalized improperly due to these three errors, and this is why some "road dog types" thoroughly mistrust measurement geeks.

An RTA (Real-Time Analyzer) is a two-dimensional measurement system that displays energy in dB SPL or volts versus frequency in hertz. Meanwhile, (Gold-Line) TEF, (SIA) SMAART, (Meyer) SIM and the like are all three-dimensional (3-D) measurement systems that display energy versus 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. When measuring 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 loudspeaker 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 loudspeaker to be flat on an RTA display with the measurement mic in the middle of the listening area, you'll be unpleasantly surprised by the resulting bad sound quality. This is not a similar measurement to the electrical one by any means.

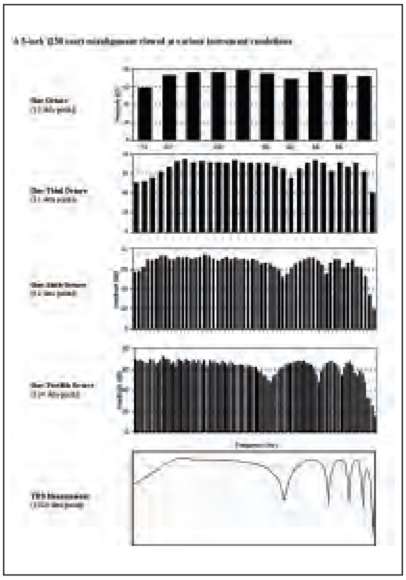

Figure 2: Various measurement-resolutions of a comb-filter caused by a 3-inch signal delay. Even the 1/3-octave resolution cannot clearly show that the frequency-response problem is non-minimum phase, time-oriented, and therefore cannot be equalized. (Courtesy of Pat Brown, Syn-Aud-Con)

TYPES OF CURVES

Probably the first person to recognize this difference was Dr. Charles 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's largely the acoustical power response of the speaker system, as modified by air and surface absorption within the room. (**Figure 1**, previous page)

What is the power response of a loudspeaker 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 loudspeaker produces.

For example, let's measure a two-way loudspeaker 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 of the high-frequency energy is aimed in the general direction of the mic due to the directional effect of a horn on the high-frequency (HF) driver. However, most of the low-frequency energy is not aimed at the mic because the low-frequency (LF) driver is omni-directional for most of its passband.

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 sound pressure level (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

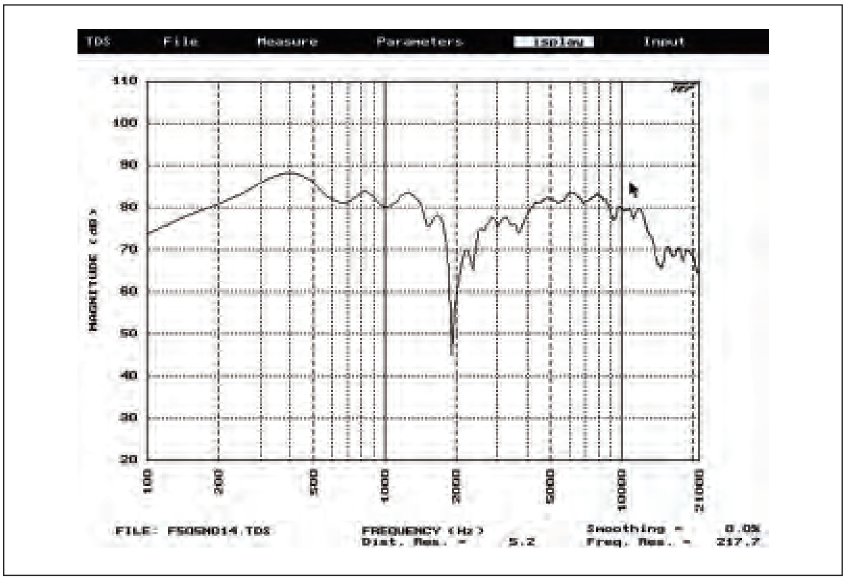

Figure 3: Non-synchronized LF and HF drivers cause the notch at 2000 Hz. Since this problem is due to the later arrival of the signal from the system woofer, it is a time-related, non-minimum-phase event, and is also not equalizable. Viewed by an RTA in the reverberant field, this notch would not be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

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 most other sound reinforcement applications, 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.

TIME ORIENTED EVENTS

Let's shift focus to "non-minimum-phase anomalies." This $10 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. (**Figure 2**, previous page)

Another is the notch at the crossover frequency of a speaker system when the drivers are not time-synchronized. (**Figure 3**) Neither of these frequency-response problems can be remedied by equalization. These are non-minimum-phase 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. Yes, I can already hear the protests to this statement: "But I've had to change the 'room EQ' numerous times when it was equalized empty early in the day, and then didn't sound right when the room filled with people because it was less 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 loudspeakers, not the reverberation. These effects on the direct sound are not time-oriented; rather, they are minimum-phase and can be equalized. Why else would a BSS Omnidrive include a meteorology probe?

Congratulations – you say you've gotten a new 3-D, FFT-based measurement system and you're going to tune? Next time we'll talk about how to do this correctly and optimally. ■

Live Sound Technical Editor John Murray is a 27-year industry veteran who has worked for several leading manufacturers. He has also presented two published AES papers, chaired four Syn-Aud-Con workshops, is a member of the TEF advisory committee and an ICIA Adjunct Faculty member. He can be reached at jmurray@livesoundint.com.