

Why Measure?

A detailed look at where it matters

By Bob Thurmond

There exists a long list of important sound system characteristics that we have no meaningful way to measure or evaluate. (*Think Tank*, April and July 2004 issues). Besides, the really important skill is to be able to listen, identify problems by ear and correct them on the spot, right?

But is this realistic or just an elusive dream?

Even if problems can be identified subjectively, they still need to be solved physically. Therefore, we must know exactly what, where and how much. These parameters can be learned with a lot of “cut and try” – or by measuring.

True, there are cases where meas-

urement did not lead to a good solution, but this doesn't mean that it's useless. Rather it's more likely that the measurements taken were inadequate, or weren't the right ones, or that the results were misinterpreted.

Meaningful, useful system performance measurements have been taken and used beneficially for decades. The techniques involved are widely available and not difficult, but still little known. Perhaps looking at a specific example to see exactly what was done, what was learned and what was gained can be instructive.

WAY-BACK MACHINE

It's 1984, and I'm working at the Aquacade, part of the New Orleans

World's Fair. The test equipment is a standard GenRad (now Teradyne) unit. Why look at a 20-year-old project done with 50-year-old test gear rather than something more current? Stay with me...

The project requires me to assist in the design and execution of systems at several venues, and nearly all of the sound system equipment has been supplied by a MI manufacturer. The design challenge: figuring out how best to use the given system items.

Program material for the Aquacade is pre-recorded stereo music tracks and live announcements via a wireless microphone system. The loudspeaker allotment is just six two-way cabinets (compression drivers on 90-degree horns plus 15-inch woofers). The seating area to be covered are fairly high bleachers gently arced around the pool. (**Figure 1**)

I succeed in getting the loudspeakers located high enough to attain decent front-to-rear coverage, but they are required to be housed inside weather-resistant enclosures behind two “corners” of the pool. (Talk about interesting resonances!) Each enclosure includes three loudspeakers, placed side-by-side and aiming straight out. Horizontal coverage stinks, and overall sound quality isn't much better.

With several other venues on my project list, time is tight. Only the most essential measurements can be taken, and topping this list is overall system response.

First, both of the outside loudspeakers in each group are angled outward



Figure 1: The Aquacade, with loudspeakers located in the dark sheds atop the structures behind the corners of the pool.

about 45 degrees in order to broaden and smooth the horizontal coverage. Then I place a flat-response omnidirectional mic at ear level in the seating area and feed pink noise through the system, one channel at a time.

The mic signal is routed to a third-octave filter, which is swept slowly and continuously over the entire frequency range, and then plotted on a graph. This process is repeated at a total of eight mic locations throughout the seating area. (Left channel results can be seen in **Figure 2**.)

Close examination shows that each trace is a different color, and each is the measured frequency response at one of the mic locations. Following an individual trace clearly shows places where the level varies up and down over a range of 5 dB or 10 dB in less than an octave. This is a very ragged response and represents what a listener at that location would hear. Such variations are quite audible – and don't sound good!

Further, looking at a particular frequency shows that the level of the different traces varies, typically by 5 dB or 10 dB as well. This is variation in level at that given pitch from one audience location to another. Also not good.

Already, there is a lesson: with such strong response variations from one location to another, no one location can even come close to providing an accurate indication of the overall system response. Experience and a few studies indicate that six samples, well spaced, are an absolute minimum. More is needed in cases like our example, where there is a strong variation from one location to another. It's likely to be the culprit of poor equalization results.

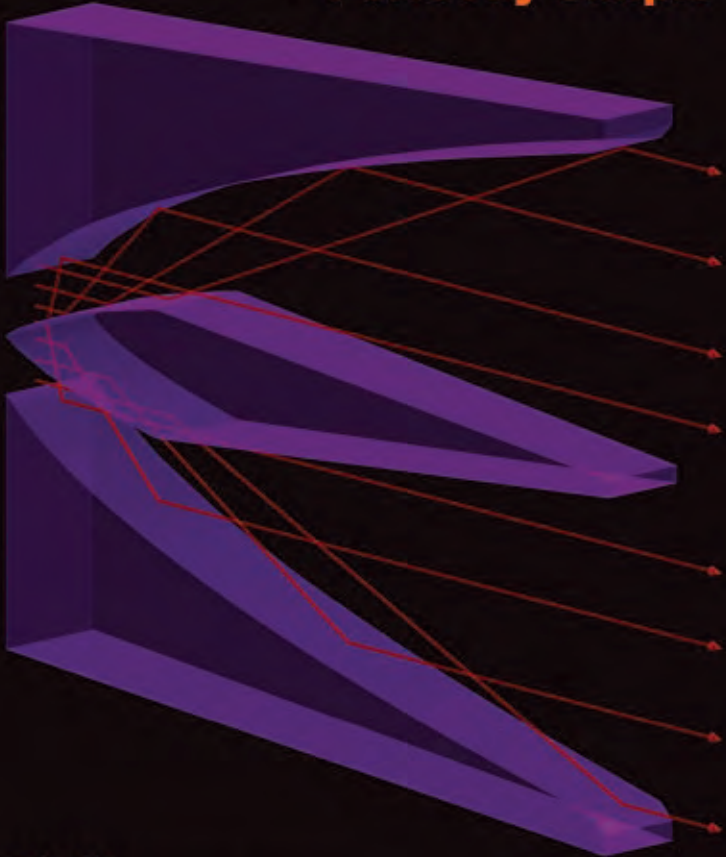
SEARCH FOR MEANING

But what does it all mean? Do these irregularities relate to some characteristic of the sound, or are they just another academic exercise? And if there is a relationship, how close is it? How much irregularity is noticeable and how much is unacceptable? What is the actual significance?

As with many sound characteristics, these questions have never been studied. This doesn't mean they're

Butterfly...

Perfectly Simple



(*) Source: Butterfly System White Paper by Guido Nozani

D.P.R.W.G.


Double Parabolic Reflective Wave Guide (*)

Single HI-Pack element features:

- Volume = 0.067 m³
- Weight (including flying hardware) = 35 kg. / 77 lb.
- Number of speakers = 5
- Centre-to-centre step = 0.243 m. / 9.5 in.
- Continuous power = 920 W RMS (AES Standard)
- Max. SPL (peak) 1 m. = 140 dB

16-element HI-Pack array features:

- Max. array height (straight array) = 3.89 m. / 12.8 ft.
- Total weight (including flying frame) = 620 kg. / 1365 lb.



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insignificant! A system with measured characteristics like these will not sound as good as it should, and the cause of the characteristics is probably destructive interference between multiple sound sources. Improving the loudspeaker configuration should improve the sound quality. (Another likely conclusion is that the measurements are at least as sensitive as our ears are at identifying the problem and indicating its severity.)

In this case, like many others, there was no opportunity to reconfigure the loudspeakers other than the slight re-aiming. The only realistic thing left to do was correcting the overall frequency response/timbre/tonal balance as much as possible.

But: exactly what was this overall characteristic? In an attempt to define it, I carefully drew an average of the various traces, as shown by the heavy line. This technique is admittedly questionable, but it does reveal some interesting information. (again, **Figure 2**)

First, it indicates that the overall response is neither flat nor smooth. Most prominently, there is a dip centered around 720 Hz and a peak centered at 4700 Hz. These do not look too severe on this stretched-out graph, but note that the difference in level between these two points is 15 dB. Quite significant! Incidentally, the loudspeaker manufacturer's data sheet indicates a similar curve, so this is believable.

There are also numerous smaller response irregularities that may also be significant. Again, this has never been thoroughly studied, but limited studies have shown that a third-octave response bump which is 2 dB or 3 dB high can be clearly audible, as can a hump an octave wide which is only 1 dB high.

Most certainly, there should be concern with irregularities of these magnitudes if they're consistent over much of the seating area. For example, the irregularities in the 400 Hz to 500 Hz region are clearly in this category.

Our ears are most sensitive to response irregularities in the region of a few kHz, and become less so as we move away from this region. The response below 100 Hz is probably acceptable, but the region between 125 Hz and 250 Hz could benefit from a little correction.

PRACTICAL LIMITS

So, in general, we want the overall measured response to be flat and smooth, but there are practical limits on this. The most important exception to this is that decades of experience, and very few relevant studies, indicate that the response should slope down gently above a few kHz. This varies somewhat with the situation, but it is almost always true that a completely flat treble response sounds unnaturally bright.

The next step, then, is to draw a desired response curve, with the high-frequency roll-off, and then draw a curve indicating the difference between the measured and desired responses. This will be the desired correction; the response of the needed equalizer. I did this on another section of graph paper, and then tuned an eight-section parametric filter set to provide an overall response, which was very close to the desired curve. (I then repeated the process for the right channel.)

The essential next step is proving the results by re-running the system response measurements with the equalizer in the system. Any remaining flaws are then tweaked out and rechecked. Again, with the time-crunch situation, I was only able to run a

quick response curve at two audience locations, the average of which looked good. The sound people wanted to hear the results right now, so I played some music appropriate to the location and held my breath.

Expressions of surprise began to appear on their faces. The director looked at me and asked, "Is this the same system?" There was even a reference to a "silk purse and a sow's ear."

I wanted to measure the response again, this time using the announcer's mic – I knew the mic's cartridge was not flat! I was told not to bother – the music quality was the important thing, and the sound crew expected to be able to tweak the sound of the announce mic with the console's three-band equalizer.

Several months later, having received no complaints, I went back, mostly just to enjoy the fair, but also to check out the sound systems, half expecting to find my equalizers bypassed or replaced by third-octave graphics. Instead, everything was as I had left it. I suspect that there were several attempts to get better results with graphics, but apparently everyone decided that my equalizers sounded better. (I didn't press the issue!)

However, I inspected the console EQ settings closely. Both music channels had only a slight boost in both the bass and treble. The live announce channel, however, showed strong settings in all three bands. Rather than try to figure out how these settings and the mic response might combine, I decided to just measure the overall result.

My flat-response mic was placed at an arbitrary location in the seating area and recorded the response with the console EQ bypassed. Then I substituted the announce mic in the same

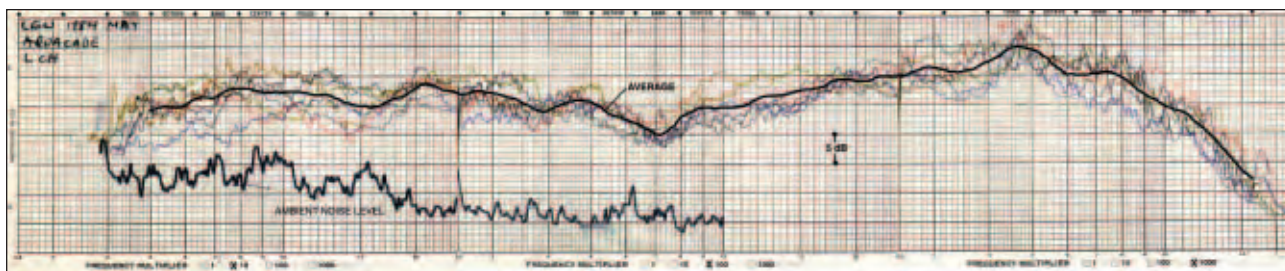


Figure 2. Third-octave frequency response of the left channel, measured in eight audience locations.

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location, aimed at the loudspeakers, with its console EQ settings restored, and ran another response. (Figure 3)

Several features are noteworthy. First, although not noted on the chart, it's apparent that I made these measurements with my equalizer bypassed, as the responses are very close to those of the original condition.

A more surprising observation, though, is how closely these two traces track together. This means that the console EQ settings nicely flattened the response of the mic cartridge. It also means that this flat response is what was perceived and preferred by the system operators. Interesting!

What does all of this teach us?

Valid measurements can yield very useful information.

This information can accurately describe the timbre/tonal balance of a system, and provides this description in at least as much detail as our ears can detect. It shows how and how much the performance deviates from the ideal, and further, reveals this at any given location, or many locations, detailing how the performance varies from one location to another.

The information also suggests specific physical causes for performance deficiencies and the means to correct or reduce them. It verifies corrective efforts and provides new information for further improvement, and distinguishes between multiple shortcomings and causes, which may be impossibly confusing to the ear.

Finally, it provides documentation that can be used for future reference or comparison. This can be of a quality

and quantity far exceeding what can be obtained just by listening. And used properly, the information significantly improves the competence and effectiveness of any sound professional.

Valid measurement results can be costly to obtain.

The necessary test equipment costs thousands of dollars, and it's surprisingly difficult to find equipment capable of running all tests needed to provide the desired information.

Significant learning and experience are necessary for competent results. It's easy to fail to obtain the needed information or to obtain incorrect or inaccurate information. Even if the correct information is obtained, it's just as easy to misinterpret it or overlook some of its significance – or to misunderstand, or fail to understand at all – what corrective actions should be taken.

Any of these mistakes can lead to incorrect actions that result in system performance that is insufficiently better, or even worse, than before. Such an experience can lead to the conclusion that measurements are worthless, and, therefore, to the failure to utilize a valuable resource.

On the other hand, using this resource can result in costly damage to cherished opinions and beliefs, and even to the artistic ego. Be warned!

Many other measurements need to be made.

While the overall frequency response is almost certainly the characteristic most important to perceived sound quality, there are many others that should not be ignored. These include maximum sound level, overall

coverage uniformity, freedom from feedback, and factors affecting reliability – all warrant extensive discussions of their own.

In addition, the characteristics examined here are not fully measured by the means discussed; new techniques are always needed. Most particularly, there is currently no realistic way to measure the nature and uniformity of the direct sound field in a reverberant environment.

And it's not at all clear that we're correctly interpreting or understanding the measurement results. Sadly, little real research has been done in this area.

However, there is one thing that's pretty clear: the flatter or smoother or more uniform the measurement results, the better the system will sound. Measure well and prosper.

Final note: I still have, and occasionally use, the GenRad gear used to make the measurements shown in this article. Even though it dates back half a century, this unit is still capable of making many important measurements accurately. But it's also large and heavy. For the past 18 years, I've employed a Neutrik Audiograph, which performs very comparable tests and is an order of magnitude smaller and lighter. I've yet to find a computer-based configuration that can adequately perform all tests I consider essential. ■

Bob Thurmond began his professional audio career in 1959 as a recording technician. He became a consultant in 1965, specializing in sound reinforcement and formed his own company, G.R. Thurmond and Associates, in 1971. Reach him at thurmond@io.com.

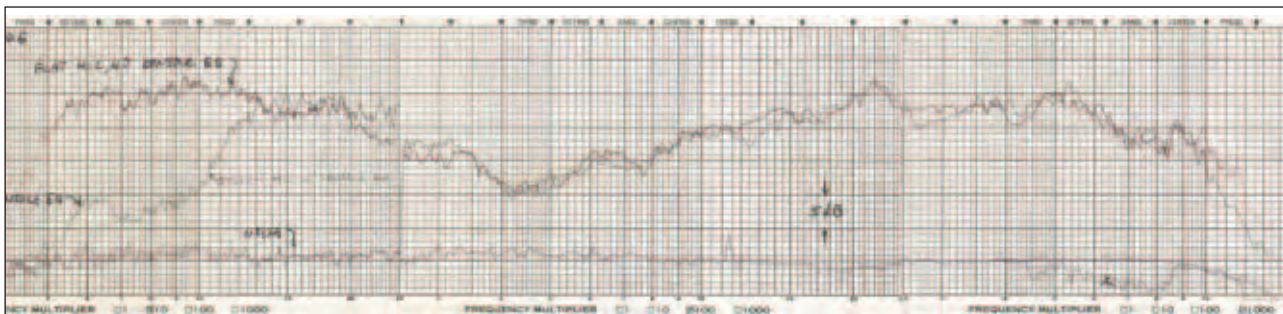


Figure 3. Frequency response measured with a flat-response microphone (with no equalization) and the announcer's microphone (with mixer equalization settings preferred by the system operators) at the same location