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MEASUREMENT BEYOND THE ATOMIC LEVEL

The setting is surreal. As you drive into the Satsop Business Park in rural Elma, Washington (pop. 3500, max), eyes immediately fixate on the looming 481'-tall cooling towers of an abandoned nuclear facility.¹ Remnants of the largest nuclear power plant construction project in the United States,² the site was mothballed in 1983, in part due to concerns triggered by reports of what had happened at Pennsylvania's Three Mile Island four years earlier.

The abandoned site's reactor and turbine buildings now house NWAA Labs, a 13-year-old independent laboratory that tests loudspeakers and materials for the audio, acoustics, and construction industries. NWAA Labs' founder is Stanford-educated electrical and mechanical engineer Ron Sauro, 76. Initially, Sauro juggled work at NASA with performing as the Vox organist of the 1963 Gold Record-earning group the Rivas.³ Sauro later became a sound



system designer/installer for churches and arenas. He began measuring speakers and contributing papers to the *Journal of the Audio Engineering Society* circa 1990 and opened his first speaker-measuring operation, Western Electro-Acoustic Labs, in 2005.

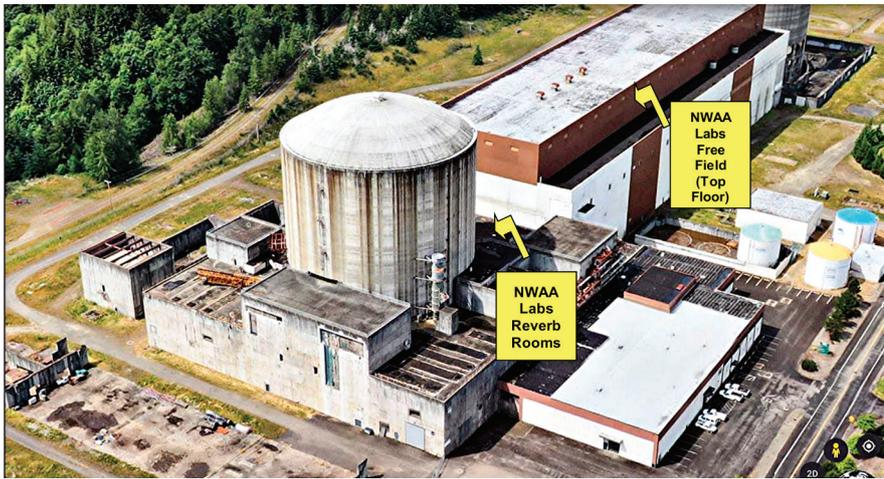
To say that NWAA Labs is unique is a grand understatement. The lab, which boasts the two largest reverberation chambers in the world, occupies a temperature-stable building designed to withstand a 10-megaton nuclear blast on its roof and a magnitude 10 earthquake. The building rests on an exceedingly stable sandstone layer roughly 10,000' thick. After excavating a 500' x 500' hole to a depth of 300', the Washington Power Supply System (WPSS) dug a 6'-wide trench around the foundation slab. By separating the foundation's sides from the earth's crust, the trench was intended to prevent earthquake damage and transmission of vibration and noise.

Sauro likens the facility's construction to three nested Russian matryoshka dolls. He describes the structure in some detail: The outer-building walls are 5' thick and contain eight layers of 3" rebar. Forty feet inside those walls, a 6" trench surrounds an inner building of identical construction. Inside this building, within another, 1'-wide trench, a 3'-thick circular containment vessel, also of concrete and rebar, was built to hold the reactor. Three nested structures were thus created, each with its own floor, ceiling, and walls. A 40'-deep depression was then excavated underneath the containment vessel and made into a water

1 See dailymail.co.uk/news/article-3158297/Eerie-images-abandoned-nuclear-power-plant-Elma-Washington.html.

2 Intended as a 3.9-gigawatt nuclear power plant that didn't need active pumps for cooling, it was on course to be the only completely passive Series 2 reactor in the United States. Instead, it was eventually given to Grays Harbor County, which turned it into a business park.

3 He also played Hammond B3 briefly with Santana and the Doobies.



reservoir. In case of a meltdown, the vessel's contents would drop into the reservoir and the fuel would solidify rather than burn.

In November 2021, I accepted an invitation from acoustical engineer Norman D. “Norm” Varney of A/V RoomService⁴ to explore Sauro’s multifloor laboratory. I joined a facility tour with members of ASA, the 7500-member Acoustical Society of America.⁵ After this event, I arranged a four-way Zoom and phone conversation with Sauro, Varney, and *Stereophile* Technical Editor John Atkinson.

When John learned of the chamber’s size and its five overlapping modes at 25Hz, he marveled at the labs’ capability to conduct extremely accurate power-response measurements.

“At very low frequencies,” Ron interjected. “NWAA Labs can measure accurately down to 25Hz because it resides in a floating room inside a floating room, separated from the outside world by roughly 25’ of concrete. The room’s background noise at 1000Hz is an astounding -43dB!” (This is in reference to a scale where 0dB = 20µPa, the threshold of human hearing—astounding, indeed!)

“It enables us to do some interesting transmission loss [TL] measurements, since we have 140dB on the other side of the TL opening between rooms,” Ron said. “We average about -17dB in this room if you measure from 25Hz to 10kHz. That gives us almost 160dB of variation for transmission-loss tests. Therefore, when we do TL measurements, we can do full measurement for all frequencies between 25Hz and 10kHz with no notations or compensations.”

NWAA Labs serves clients worldwide, with about half of its business coming from Canada (whose main measurement lab is

Clockwise from top: This converted nuclear power plant is now used for measuring loudspeakers; ground view showing the Free Field Lab and the Reverb Room; Norm Varney in front of the microphone array.

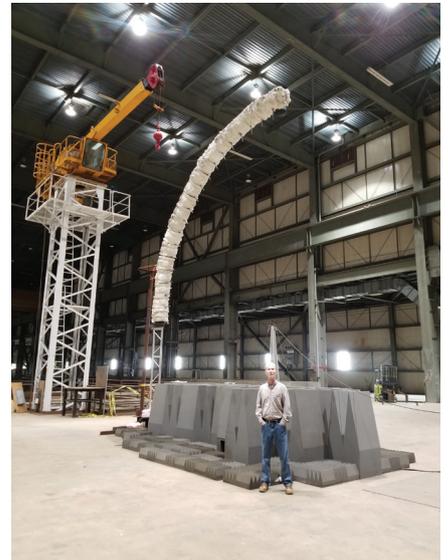
the National Research Council, or NRC, in Ottawa). Anechoic chambers, which are used to simulate a *free-field* environment devoid of reflections,⁶ are few in North America: Paradigm has one in Toronto; Boeing’s, the largest in the US, is 65’ × 65’ × 65’.⁷

NWAA Labs’ “turbine room,” which is 650’ long, 350’ wide, and 80’ high—large enough to hold four NFL football fields—is a true free-field space. The roof area is a whole acre. As John noted, it’s so large that there’s no need to gate impulse responses in this space because reflections are so attenuated after 160ms.

John was fascinated by the room’s huge, curved microphone array, which contains 19 matched Earthworks M30 (ANSI Type 1) measurement mikes. The array is 4.1m in radius and is suspended 15’ above the floor. Microphones are mounted within 1/16” of a central point, one every five degrees. Each mike is electronically compensated for distance to one sample at 48kHz.

The diffuser Ron was measuring at the time of our interview sat on a pedestal connected to an LS-360 turntable that could move in increments of one-tenth of a degree. At the mike array’s typical distance of 4.1m from whatever is being measured, Ron can accurately measure phase response at 10Hz.

John asked whether at a 4.1m distance it was possible to measure a large speaker and get proper integration of drive-unit outputs. Ron replied, “We know what our limits are. The spacing of drivers cannot exceed 48”.” John, who acknowledged his



envy, replied, “The big problem with loudspeaker measurements in general is you need to be further away than the largest dimension of the loudspeaker. That’s fine for an LS3/5a but impossible for a significantly larger loudspeaker.” Ron responded, “We can measure subs all the way down to 25Hz when we support them above our 5’-tall wedges. The wedges you see around the measurement pedestal prevent reflections from the floor. Our biggest sub has been something like 570–600lb. We also do long line arrays with different cabinets, as in a PA system. We use a forklift to drop heavy speakers into place.”

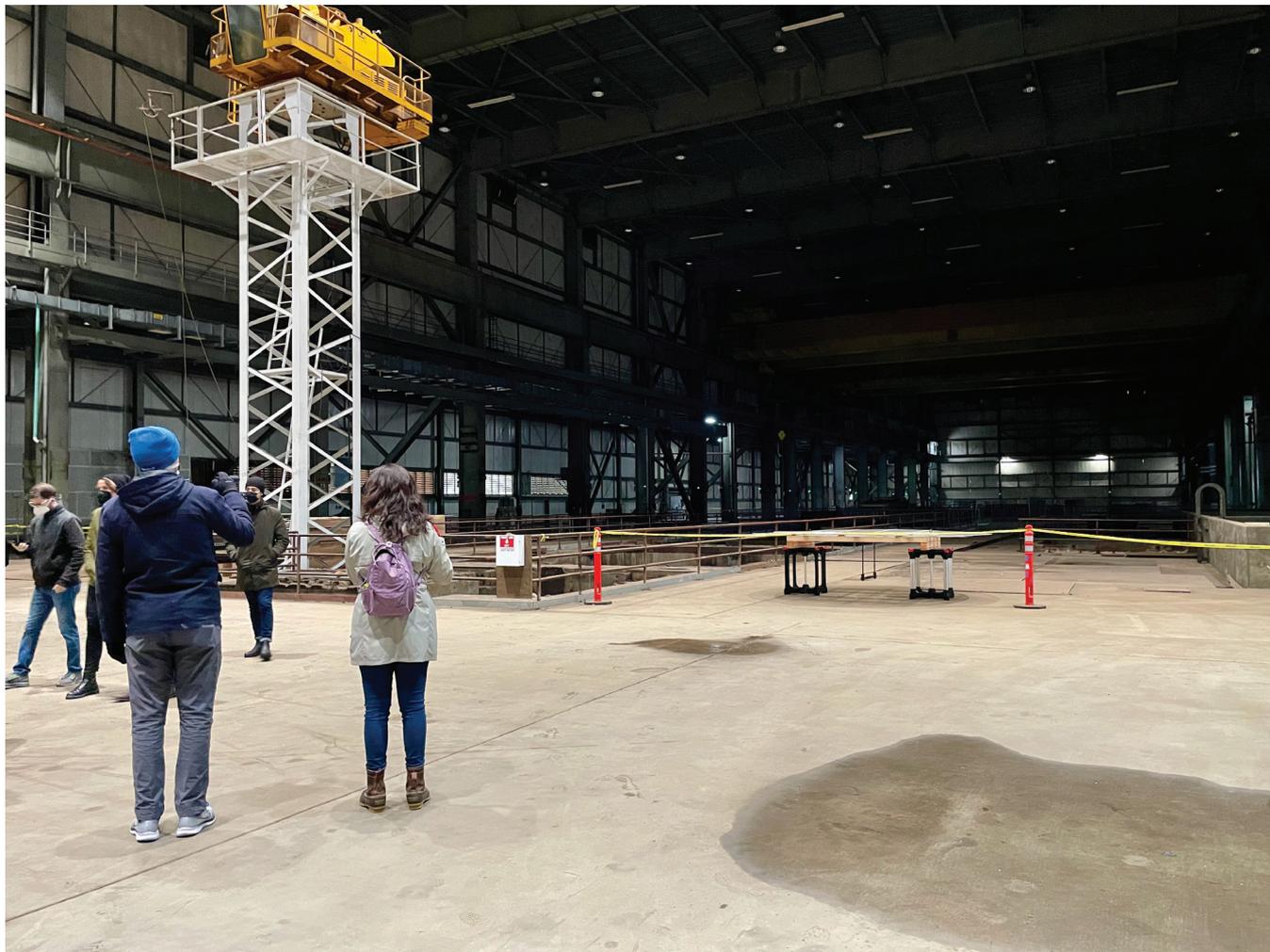
When NWAA Labs opened, they focused mainly on pro-speaker measurements. As time progressed, they also began measuring audiophile brands including Bowers & Wilkins and Genelec. “You

⁴ Varney has used NWAA Labs for testing over the years.

⁵ See <https://youtu.be/JDyvmgMCuNY>.

⁶ “Anechoic,” in Greek, means free from echo.

⁷ The famous one at Bell Labs’ Murray Hill, New Jersey, facility—now part of Nokia—where much important science was done, is now used to create art. See youtu.be/wqCjE2WzhBk.



name it, we've done it," Ron said. "I've measured approximately 300 brands—probably a little over 3000 speakers—over the last 10 years. Speaker measurement is probably about 20% of our business."

Loudspeaker measurement

After acknowledging NWAA's fantastic ability to measure the far-field directivity, magnitude, and time-domain response of the device under test, John asked, "In your opinion, Ron, which of the measurements best correlates with what people hear? When they say that a speaker sounds 'good,' are they talking about the flatness of response, the evenness of dispersion, the coherence of the time-domain presentation, or the absence of distortion and resonance? Which is most important? My experience with subjective-listening tests is that people will agree if a loudspeaker has elevated low frequencies, but half will say that's a bad thing and half will say it's a

good thing. They agree on the perception but disagree on the value judgment."

Ron offered no simple answer. "AES, ASTM, and all the groups that help develop standards have been working for 25 years to come up with a standard that says, 'This sounds good,' and we have not even come close to agreeing. All we can do is objectively describe what we can measure in terms of frequency response, phase response, radiation patterns—the things that don't change, depending upon the flavor of the speaker."

Norm, whose A/V RoomService focuses on resonance and acoustics, noted that even though all currently measurable aspects of speaker response interact with each other to produce the sound that reaches our ears, acousticians typically measure speaker response by playing a swept sinewave or pink noise. After that, they examine one parameter at a time. Our ear-brain system, however, performs multiple calculations

NWAA's free-field room is the size of four football fields—so large that there's no need to gate impulse-response measurements.

simultaneously, accesses our experience, and then passes value judgments.

"This is why you have to spend some time learning how to look at the different measurements and determine how they all integrate with each other," Ron said. "You have to integrate them in your brain and understand that if you take this particular parameter and combine it with this other parameter, it's going to give this type of result. That takes time and experience. You can't just throw a graph out there and expect that somebody who has absolutely no experience integrating these measurements will understand what they're seeing."

John agreed. "As I wrote years ago, to understand measurements, you have to look at all of the measurements simultaneously. In the measurement sidebars in

Stereophile, I try to characterize what is happening in ways similar to what you guys are describing and then look at measurements which are so out of the ordinary that they may have an effect on sound character.

“One of the things I fall over all the time is the effect of resonances. Floyd Toole and Sean Olive have said that the higher the Q⁸ and the higher the frequency, the less likely the resonance will be excited and the less likely that it will affect sound quality. I find that some resonances have a low enough Q and a low enough frequency that I hear them when I’m doing the measurements. Yet, other people out there don’t mention the resonance-induced coloration that I’m

hearing.”

Ron replied that each frequency range has a different driver set and a different far-field point. This distance, called *critical distance*, is where the *direct sound* from the speaker is equal in magnitude to the sound of the *reverberant field*.

John shared an anecdote. “Many, many years ago, I visited the home of the late Alastair Robertson-Aikman, the founder of SME, who had a very big listening room—something like 35' × 25', with a 15' ceiling. He equalized the sound at the listening position to be flat. But the highs were so exaggerated—tilted up like crazy—because he was equalizing the reverberant field.”

“There are so many people who think they’re experts,” Ron lamented. “They take a Radio Shack level meter and say, ‘See, it measures equally all the way around, so we have good coverage.’ I look at them and say, ‘No, you just happen to be measuring the reverberant field far enough out to where it won’t show up all your mistakes.’ ... The only way a speaker owner can properly use measurement for setup is to sit within the area where the direct sound of the speaker reaches them, and sound isn’t affected by the reverberant field.”

⁸ John Atkinson says, “Q stands for Quality Factor—it is related to the amplitude of a resonance and the frequency range that it covers.”

NWA'S RON SAURO ON SHAPE AND ABSORPTION

Our conversation moved back and forth between absorption and diffusion. When, later, Ron attempted to separate the two, he produced a mini treatise on absorption. Lightly edited, that treatise follows.

Absorption is the most common treatment used in listening rooms. The measurement of absorption was first described by Wallace Sabine, who compared the reverberation times of a room with and without absorption. This difference was then converted to units of absorption using this formula:

$$A = 0.9210(V \cdot d / c)$$

where

A = equivalent absorption area in m²,

V = volume of reverberation room in m³,

c = speed of sound at ambient temperature and humidity in m/s,

and

d = decay rate in dB/s.

Most of what we think we know about absorption is wrong! Absorption calculations in use today can err by as much as 85%. The biggest error concerns how important the area of absorption is in the calculation of how much absorption is needed.

Area is not the only factor that affects absorption. In standards used by either ISO (International Organization for Standardization) or ASTM (American Society for Testing and Materials), when absorbers are measured in reverberation rooms, the edges of the samples are covered with a material, such as painted wood or sheet metal, that is impervious to sound. The assembly is taped

to the edge coverings and the coverings taped to the floor. This method removes the edge thickness from the calculation of the absorption coefficient. When this method of converting the amount of absorption into a constant was first proposed at an ASA meeting in 1927, it was questioned. But no one proposed an alternative method. The formula was a very simple:

$$a = (A_2 - A_1) / S + a_1$$

where

a = absorption coefficient of the specimen,

A₁ = absorption of the empty reverberation room,

A₂ = absorption of the room after the specimen is installed,

S = surface area of the specimen, and

a₁ = absorption coefficient of the surface covered by the specimen.

Under the presupposition that all numbers resulting from this calculation will fall between 0 and 1, this theoretical absorption constant was then described as a mathematical coefficient.

Imagine the shocked looks when results exceeded 1. After the first attempt at an explanation failed, most agreed this was an effect of diffraction of the edge and of little consequence. The phenomenon was then renamed *edge effect* and invoked as the reason for numbers that did not add up mathematically. Even though no one could explain what edge effect was, they blamed it for the “bad” numbers.

As part of the process of measuring

scattering coefficient, acousticians used the ISO 17497-2 method to determine the absorption of a *circular* sample of X area and convert it to an absorption coefficient. Since there is also a different standard ISO method, ISO 354, to measure the absorption of a *rectangular* sample of the same area, we at NWA Labs thought it might be useful to compare the results of the two methods. When we did, the circular sample measured about 15%–20% less than the rectangular sample, even though they both covered the same area. We repeated the test many times and got the same results.

Next, we began a series of tests where we broke up the area into small segments in different configurations. Our results produced an ever-increasing absorption coefficient. When these tests were plotted on a graph as result vs area, the results were linear to a point where it was 99.9% certain that the ratio of linear edge length to area was the operative denominator in the coefficient calculation. Reports of these experiments in Institute of Noise Control Engineering and ASA papers exposed the ineffectiveness of the absorption coefficient as a true constant for calculations of absorption. Later papers showed that calculation to be off by as much as 85%.

You can change absorption by up to 80% just by changing the absorber’s shape. If you take a piece of material, let’s say 100 square feet, and you make it into a circle, it will provide the minimum amount of absorption. If you fashion it into a square, it will be just slightly more absorptive. But if you make it into a star shape, you get almost 100% more absorption out of the same piece of material. —RON SAURO

Drowning in sound

“When you’re listening to sound in a room, there are two different kinds of sound,” Ron said. “If you’re within the critical distance from the speaker, you will hear the direct sound of the speaker. If you’re beyond the critical distance, you will hear the reverberant field, aka the soundfield that is developed by the room.

“The best way I can describe the room’s soundfield is by analogy. Let’s pretend that a bathtub is the room, the bathtub drain is the equivalent of the absorption in the room, the input pipe is the speaker or sound source that adds energy to the room, the water is the reverberant soundfield, and the air is the direct soundfield.

“If you’re sitting in a bathtub with a 1” drain and a 3” input pipe, how long before the bathtub fills up? You’ve got to think about this for a second, because the drain is releasing water slower than it’s coming in. What happens, where? Does it overflow at some point?”

Without hesitation, John responded, “It’s going to reach an equilibrium point when the height of the water forces the water out of the 1” drain as fast as it is coming in from the 3” pipe.”

“Bingo,” replied Ron. “The pressure is going to change. It’s the one thing that’s not shown. As you increase the level of water, you’ll increase the pressure, and the pressure will increase the flow rate. It will stabilize at the point where those all equal out.

“But here’s the problem. Let’s say you’re lying back in this bathtub full of water, and you slowly but surely slip down into the water as you relax. Your nose gets closer and closer to the level of water, to the point where the water reaches your chin but your nose (ears) is in the air (and your ears are in the direct field). Then you keep sinking into the water until it has finally covered your nose and mouth. If you try to breathe (listen), you will drown regardless of how far down your nose is.

“The water is the reverberant field. The depth of your nose in the water is the equivalent of how far back you are in the reverberant field.

“This is what listening is. The reverberant field in a room is equivalent to the level of the water of the tub. Therefore, if you’re in the back of the room, you are in the reverberant field, and you’re hearing the same level as if you were under water.”

“But that would mean you’re beyond the critical distance in the room,” objected John. “You’re sitting a long way away.”

“If you get beyond that point, you do not



Norm Varney of A/V RoomService stands next to a large acoustic panel under test.

hear the speaker at all,” Ron responded. “All you hear is the room.”

Diffuser measurement and evaluation

NWAA Labs has been researching diffusion for the last five or six years and is developing a measurement standard for ASTM. Ron claims he can often tell whether a diffusion panel is effective or worthless just by looking at it.

“Because we look at diffusers as though they are a speaker without a voice-coil, we use the same measurement system for both products. At the very top of the microphone arc, behind the microphones and slightly off to the side, is a speaker. The speaker enables us to ‘illuminate’ the diffuser beneath it with sound waves so we can measure the first-order reflections and determine what’s going on.

“We use diffusers in simulation programs exactly as we do a speaker. We can actually group them together and form clusters and then come up with a combined radiation pattern using convolution techniques. We can create 360° plots to determine how soundwaves hit a spot on a wall and radiate off it.”

When questioned about the efficacy of the various diffusion panels, some fancifully constructed, that are marketed to the audiophile community, Ron touted the work of Virginia-based Jim DeGrandis of Acoustics First Corporation. Ron has co-authored a paper with DeGrandis (and with Dr. Hassan Azad) on using laboratory measurements to improve acoustic simulations. Both are active in the ASA. Some of DeGrandis’s papers, as well as many by Sauro, are linked to on the NWAA Labs homepage.

“DeGrandis has done some unbelievable research in the field of diffusion,” Ron said. “He’s come up with a computer program that allows him to simulate what happens to a design when he changes parameters.

“We’ve done the same research with diffusion, where shape is again key. I’m sorry, but almost 90% of what’s out there, theory-wise, is BS. For example, you can’t use a block’s length in a ‘skyline-style’ diffuser to determine the frequency range that it affects.”

It goes without saying that the next question from both John’s lips and mine concerned the efficacy of audiophile-approved devices. Which of them *do* work correctly?

“Let me put it this way,” said Ron with circumspection. “Devices that are based on diffraction scramble the phase. When you scramble the phase, you lose localization. All the things you see, like one- or two-dimensional Quadratic Residue Diffusers, square blocks, and ‘cityscape’-type diffusers, are diffractive devices. If you don’t mind losing localization, you can use cityscape diffusers behind the recording or listening area, but they are extremely restricted in the frequencies they affect. Most are restricted to frequencies around about 4kHz and 16kHz. They do absolutely nothing below 4kHz.

“Believe it or not, if you want something that will actually move the sound where you want it to, some of the oldest devices out there are still the best. One of those is a polycylindrical or cylindrical unit. In my opinion, I would recommend sections of spheres. You take a sphere and just cut



across the chord and stick it on the wall. That's probably your best bet for doing diffusion."

John raised the issue of specular reflections—the mirrorlike reflection of sound waves from a surface—and pointed out that when that reflection arrives at the listening position, the delay between the direct sound and the arrival of the reflection, which has taken a longer path, means that it won't be in phase.

"You've got to think about far field versus near field," Ron said.

John continued, "I always understood that specular reflections are a bad thing because at the listening position, you want to hear the direct sound. All the other energy in the room should be in the reverberant field, where it doesn't interfere with the imaging. Yet what you are saying is you want a specular reflection because the sound will arrive with the correct phase relationship."

Sauro responded that the reflection will arrive with intact phase and retain localization. He emphasized that if you use a cylindrical reflector, the reflection reaches the listener's ears from only one point on the cylinder. Because reflections from all

the other points do not reach the ears, phase differential is inaudible. The amount of energy reaching the ears may be tiny compared to the total amount of sound reflected by the cylinder, but it maintains localization.

John asked whether a conventional Schroeder-type diffuser was restricted in its low-frequency performance.

"You have to look at the size of the diffuser," Ron replied. "Let's say it's 2' × 2'. If you're farther than 4' away, it does virtually no good because it functions as a point source with a single point of reflection. 2' × 2' looks like a single plane at 500Hz. That is the specular reflection. At 1000Hz, it's barely functional."

"Small room acoustics,⁹ which applies to almost all audiophile rooms, can be measured for reverberation," Norm noted. "You want linear reverberation times across as broad an audible spectrum as possible. For critical listening, we are trying to achieve an environment with reverberation times along the lines of about 0.25 to 0.4s from about 125Hz on up, and a little longer tail below 125Hz. If we have too short of reverberations time below 125Hz, it's uncomfortable, because our eyes are not

The floor and platform below the microphone array. The floor is lined with 5' sound-absorbing wedges.

making sense of what our ears hear."

Ron noted that Norm's A/V RoomService PolyFlex diffusion panels are among many that NWAA Labs has determined to work effectively in small listening environments.

For more information on absorption and diffusion, consult the papers hosted on the front page of Ron's website.¹⁰ For readers concerned with treating their own rooms, Ron recommends the first two papers, which discuss room acoustics and cost-effective ways to increase absorption.

Final thoughts and considerations

When I asked everyone to condense their knowledge about loudspeakers into a few simple bits of advice, John replied, "You have to take all the measurements and integrate them in your head.¹¹ Or you could

⁹ Smaller than approx. 15' × 15' × 15'. My listening room, which is 16' × 20' × 9', qualifies as a small room.

¹⁰ See nwaalabs.ipower.com/Papers%20and%20Publications.html.

¹¹ See stereophile.com/features/99/index.html, stereophile.com/features/100/index.html, and stereophile.com/features/103/index.html.

read the 15,000-word paper I presented at AES in 1997.”

Ron added, “You also have to keep in mind what the purpose of the speaker is. Audiophiles want a speaker that has the most even coverage possible at all frequencies.”

Just when I thought we had a wrap, Norm returned to that little omnipresent monkey wrench: the room. “Audio reviewers have an incredibly difficult challenge because they are trying to correlate bench-test instrumentation data with subjective evaluations, which, to a huge extent, are predicated on their listening environment. The speaker, the distance, the size of the room, the reverberation times, and the resonances of the room all have a huge impact on the subjective perception of the sound. There’s no way you can look at any single parameter and go, ‘Oh, that’s the way the speaker sounds.’ Not in our lifetimes, most probably.”

Ron replied, “In my segment of the industry, we say that the most important part of a speaker is the room. A good room can make a really bad speaker sound good, and a really bad room can make a great speaker sound horrible.”

John smiled. “I wish I’d said that.”

Ron affirmed that the reason he got into pure acoustics is that he figured out that if he could fix the room, he could fix most things. If he couldn’t fix the room, he had very little chance of making a bad-sounding speaker sound good.

He then recommended an exercise that *Stereophile* founder J. Gordon Holt presented in the 1970s. “If you want to know how good your speaker will sound in your room, go where your speaker is and speak—one person to another person. That is the best that sound will ever be in your room because your voice is a point source coming directly from the mouth and the room has no effect on vocal production. How the room affects the sound after it leaves your mouth is the best your voice will ever sound in the room. When we approach large rooms, we always tell people, ‘Just go to the stage and talk as loudly as you can to the people in the back row. If they can hear and understand you, that’s the best the room will ever sound. Everything else will be downhill.’” Cheery thought.

Referencing theater acoustics, Ron said, “People assume that if you have a balcony or something out in front, that you can absorb the sound by putting absorption material on it. Nothing could be farther from the truth. The best the material can do is reduce the sound of that balcony face by about 10dB.¹² That means if there



is a time delay between when someone speaks or sings and when the sound returns to them, they will hear a direct echo that will cause them not to be able to sing well unless they tune the echo out. You can determine how well speakers “sing” just by singing into a microphone, hooking it up to a 20ms delay, and then listening back through headphones.

“Small-room acoustics are different. If you are slightly out of phase, you will change the timbre of the sound, the attack, the decay, the articulation, the low-level detail, the spatiality, and more. The sound will be fuzzy.”

“If you’re building a small room for audiophile reproduction, you’ll want a speaker that doesn’t have a lot of Q,” Ron said. “You also have to set everything up for the spot where you’ll sit because you can’t set the room up so that everybody equally enjoys good sound. Pick one or two spots where you’re going to listen and make certain your first-order reflections off the sidewalls and ceiling are covered. I would recommend a spherical or a cylindrical reflector at first-order reflection points on both sidewalls and on the ceiling. That’s my personal preference because I like sound that is phase correct. I want to know where the localization is—where the singer is, and where the drummer is.

“Of course, I know recording engineers who prefer it the other way around. They prefer *not* having the localization and instead opt for an enveloping sound. That’s their preference.”

Norm suggested that there should be a standard for audio reviewers that includes good critical-listening environments in which room modes are distributed evenly

and the sound of first-order reflections is lowered. Whether this is accomplished by absorption or diffusion is less critical than keeping reverberation times linear. If you’ve got a system that has a resonance here, a boost there, a cancellation here, and so on, you don’t have a neutral reference.

According to both men, if you want to increase absorption using 1"- or 2"-inch-thick fiberglass panels, you can double absorption by spacing multiple 2 × 2 panels on the wall rather than using a single sheet of 4 × 8. “Instead of having to buy three pieces of 4 × 8, you could buy one piece, cut it up into four 2 × 2 pieces, and get the same effect.”

John said his ideal speaker setup involved having diffractive surfaces at the early reflection points and sitting close enough to ensure that direct output from the loudspeakers was at a higher level than the reverberant field. He described this as “sitting inside the critical distance.” Norm described it as “reducing the specular reflection.” John said he pays special attention to loudspeaker stereo-imaging capabilities and uses “mono pink noise” to ensure “a very narrow, stable image at all frequencies.”

Ron may prefer direct sound with direct reflections, but both men consider it essential to understand the taste of the recording engineer because it determines how their recordings should sound. Which leads to the reason many of us devote so much time, energy, and capital to equipment and setup: We crave systems that faithfully reproduce the musical experience that recording engineers (and the artists they serve) want us to hear and feel. ■

¹² Varney has attenuated ceiling reflections by 19.6dB using his FRP Ceiling Treatment.