MUSIC IN THE ROUND

BY KALMAN RUBINSON

THIS ISSUE: A visit to Tom Caulfield, two multichannel volume controls, and the 64-bit JRiver app.

Multichannel Analog Accessories

I have not been attending audio shows as often as I used to, and this January, for the first time in more than 20 years, I'm skipping the annual Consumer Electronics Show. My personal return on investment has become hard to justify, especially when attendance at each annual CES requires a round trip from New York City to Las Vegas, Nevada. More important, audio shows now seem focused mostly on either two-channel music playback or multichannel home theater, whereas what interests me is listening to music in surround sound. Sure, I can be excited by the introduction of new speakers and new power amplifiers, which have obvious application in any music system. On the other hand, the vast majority of analog preamplifiers and DACs are two-channel only, while preamplifier-processors and audio/video processors emphasize their video facilities and such sound options as Dolby Atmos, DTS:X, and Auro-3D, none of which has much impact on music.

And in disc players, there's a widening gap between the stereo-only audiophile market and the larger, HDMI-only home-theater market—and that leaves me and Oppo Digital alone in the middle. Most servers, too, seem to serve up mostly stereo.

This is not a happy situation for me, because audio shows, both national and local, give me valuable opportunities to hear good sound and music, often provided by experienced professionals. Every listener adapts to the sound of that listener's system, dominated as it is by the acoustic of that listener's room, and it's okay to accept and enjoy the resulting experience of music without constantly questioning it. On the other hand, if you're committed to optimizing your listening experience—and if you're reading this magazine, you probably are—you need to periodically refresh your ears and challenge the sound you've grown accustomed to by hearing other systems. Not all show demos are satisfying or useful, but many are interesting and refreshing, and it is such experiences that can trigger a serious reassessment of your own system.

I always attend the frequent manufacturer-sponsored events at Innovative Audio, a Manhattan dealer, and as I write this I look forward to two major events here: the Audio Engineering Society Convention (October 18–21) and the New York Audio Show (November 10–12). Still, the most valuable opportunities are to visit kindred spirits and hear what can be called a *curated* system, as in this definition of the word I recently found on Dictionary.com: "to pull together, sift through, and select for presentation, as music or website content." A perfect example was when, on my way to a Bowers & Wilkins press event in Boston last summer, I visited with Tom Caulfield.

Not all show demos are satisfying or useful, but many are interesting and refreshing.

Tom Caulfield is a Grammy-winning recording and mastering engineer who has been responsible for many of the outstanding multichannel recordings released by Chandos, Channel Classics, and Yarlung. In addition, he's a tireless champion of multichannel DSD with minimal processing. For some years now, Caulfield has also been seducing many in the audio press and elsewhere with some of his remarkable private recordings, made during commercial recording sessions but using his own microphones and mike placements.[1] I believe that he helps many of us keep a clear ear on the cutting edge of modern recording technology.

While I've enjoyed meeting Caulfield several times at industry events, as well as at one of his recording sessions, this was our first opportunity to talk at length, and for me to hear what music sounds like through his undoubtedly unique mastering system. The system dominates an upstairs room in his house, where five large Sound Lab Majestic 645 full-range electrostatic speakers, each 75" high by 34.5" wide by 7.5" deep and weighing 140 lbs, cluster closely around a comfy stuffed armchair. Together, they form an almost pentagonal enclosure in the center of the room, but because of their proximity and the fact that each speaker radiates sound from its entire surface of 1790 square inches, they "disappear" acoustically, providing a nearly transparent window on the music. There is some other furniture in the room, as well as acoustical treatments and spare Sound Lab panels, but again, with the proximity of the speakers to the listening position and the limited amount of reflected rear radiation that can reach into the enclosure, the room acoustics only minimally affect the sound. Caulfield played files from his server, which he accesses via his mastering software and routes to five Parasound Halo JC 1 monoblocks, one 400W amp per speaker, via a set of relays that permits assessment of individual channels, combinations of channels, or the entire 5.0-channel array.

I'd brought a number of familiar files with me, and Caulfield's server contained many more. Sitting in this sound capsule and with all my other senses isolated from external distractions, I felt an immediate connection to the music, and was aware of no spatial discontinuities. I could easily perceive subtleties of width and depth in the soundstage, and hear extended, detailed treble completely devoid of highlighting or brightness. Comparing really good commercial recordings, all high-resolution and 5.0-channel, with some of Caulfield's unreleased recordings using his own mikes and placements, I entirely appreciated his preference for closer miking, for a closer perspective on the performance.

1 You can get a taste of this by downloading a copy of a session outtake of the first movement of Mahler's Symphony 3, with Iván Fischer conducting the Budapest Festival Orchestra. If you can play it, select the multichannel DSD256 option; it's the unedited original made with an optimized 5.0-channel microphone setup: https://justlisten.nativedsd.com/albums/JLBFOMahler3-mahler-symphony-no-3-in-d-minor.

There was an exhilarating intimacy to the performances, even more than I'd realized when I played them at home. I also appreciated why Caulfield's system is so marvelously suitable for his mastering work. No detail is lost, and the crucial relationships among the channels is surgically revealed. That my experience was thrilling should be no surprise—these recordings were mastered on this very system. How could they possibly sound better anywhere else? This experience, and Caulfield's conversation and hospitality, affected me deeply—and left me impatient with Bowers & Wilkins's two-channel demonstration the following day.

Back home in Manhattan, it was clear to me that I would not like to live in the hothouse of Tom Caulfield's sound capsule. With its gripping intimacy came a proximity that seemed to intrude on my personal space, especially with his closely miked private recordings. I didn't feel comfortable listening from the conductor's podium (I did try it once)—the excitement simply overwhelmed the attention I was trying to pay the music. I am a listener, not a performer, and Caulfield's recordings sound more realistic to me through my own system. Still, my time with him raised my listening to a new level of sensitivity—which I will now apply as I focus on some imperfections in my system.

SPL VOLUME 8 AND SMC 7.1

It's no secret that there are very few analog control options—for volume and input selection, primarily—for multichannel, but that doesn't mean there are none. Parasound's excellent Halo P 7 is the only analog multichannel (7.1) preamplifier currently on the market, but since there has been no relevant new technology since the first round of multichannel analog preamps from the first decade of this century, one can consider any number of good used products from the multichannel exuberance of more than a decade ago. I own an Audio Research MP1, and there are others worth seeking out from Bel Canto Design, McIntosh Laboratory, and McCormack Audio. Fundamentally, all that are needed are input switching, volume control, and, perhaps, the ability to balance channel levels. Old hat.

The studio/professional market does offer some analog multichannel devices. There are lots of complex devices that incorporate multichannel switchers, analog input and output, and analog volume control, along with a plethora of mixing features, but finding something with the appropriate feature set and cosmetics is difficult. About a year ago, I discovered a device from Sound Performance Labs (SPL): the Model 2489 Surround Monitor Controller, which supports two eight-channel inputs (one balanced), one stereo input (unbalanced), and one eight-channel output (balanced). It also has a volume control with a discrete (ie, each channel is on its own physical deck), six-layer potentiometer. Unfortunately, the 2489's single balanced input connector is a DB25, its output jacks accept only ¼" TRS-wired phone plugs, and I found its looks decidedly unappealing. I let it pass.

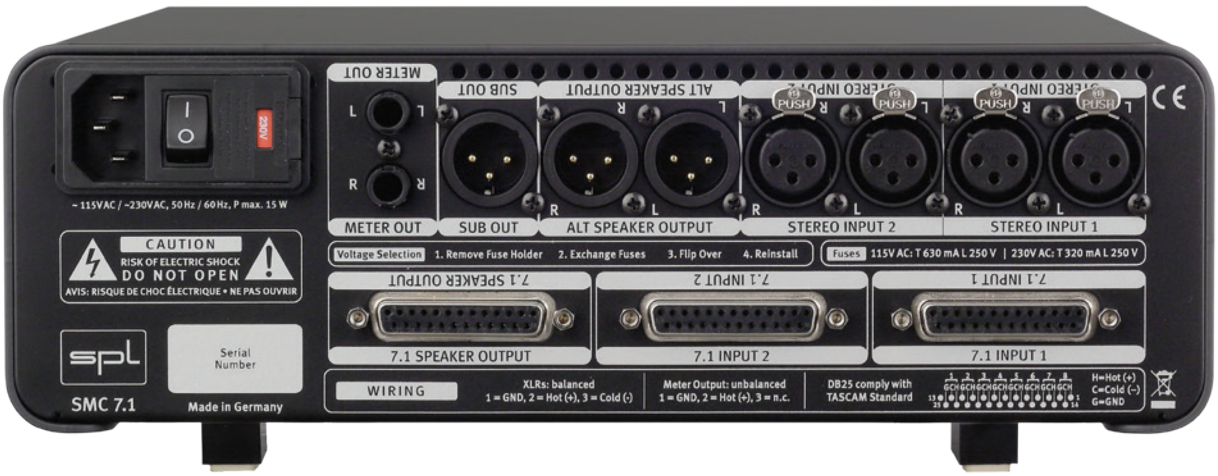
Recently, SPL introduced an array of new products aimed at getting the German company beyond the pro-sound market and into the homes of tech-savvy audiophiles. physical volume control to the output of a multichannel DAC. The first is the Volume 8 ($699),[2] which incorporates SPL's discrete six-layer potentiometer

in a neat black enclosure with an eight-channel balanced input and an eight-channel balanced output: a volume control in a box. The other is the SMC 7.1 Surround Monitor Controller ($1899).[3] It adds to the Volume 8 a second eight-channel balanced input, two pairs of XLR stereo inputs, a stereo XLR output, an XLR subwoofer output (a full-range mono sum of the L/R stereo inputs), a headphone jack, two outputs for metering, and an array of illuminated pushbuttons. Both the Volume 8 and SMC 7.1 have Mute buttons. Hooking up either requires the purchase of two DB25 cables, sometimes described as fantails, which break out into eight individual XLR connectors as input or output. Audio Plus Services, SPL's US distributor, sent along two 1m-long, multipair Sonorus Muco DB25 cables made by the Swiss company Vovox.[4] The DB25-to-XLRx8 male and XLRx8-to-DB25 female cables ($699 each) are beautifully constructed, with solid-core copper conductors for all three lines (positive, negative, and ground), and don't rely on a braided shield for the ground. And despite consisting of

2 See https://spl.info/en/products/monitor-controller/volume-8/overview.html.

3 See https://spl.info/en/products/monitor-controller/smc-71-surround-monitor-controller/overview.html.

4 See www.vovox.ch/en/professional-audio/products/sonorus/multipair-cable.



eight balanced lines each, they are also remarkably flexible.

I began with the Volume 8 and found that connecting it with the DB25 cables was much easier than I'd expected. If you attach the DB25s to the back of the component before inserting it in the rack, the multiple XLRs are now at the end of flexible links which are easier to access than the traditional connections at the rear of a chassis in an equipment rack. One DB25 was connected to my line-level XLR output cables and the other to the outputs of the exaSound e38 DAC via Hosa RCA-to-XLR adapter. When I connected the SMC 7.1, it was even easier—all I had to do was transfer to it the two DB25s from the Volume 8. It all went smoothly, and when I powered up, I heard no background noise.

Before telling you what the two SPL boxes sounded like, I'll tell you that they sounded identical to each other. That wasn't surprising, once I'd examined their circuitry and SPL's circuit diagrams. Apparently, in addition to the common discrete attenuator, both use the same eight independent input buffers and eight output buffers, each with local power-filter capacitors adjacent to each buffer board. The SMC 7.1 simply adds switching for the headphone output, more inputs and outputs, and the ability to listen independently to any channel or combination of channels.

After some careful listening with the Audio Research MP1, to refresh my ears as to the pre-SPL sound of my system, I installed the Volume 8. I was disappointed: My overall impression was of a dim, claustrophobic sound. It was a bit disturbing to hear that a "professional" device like so many with which our beloved recordings are monitored and/or processed was of a poorer sound quality than audiophiles expect. There was no air, no space, no sparkle. I listened for the rest of the day and gradually grew less bothered, but I put that up to adaptation.

There were now fewer clues to the physical presence of each of my front and surround speakers.

DID I REALLY PREFER SPL'S PRO DEVICES TO MY AUDIOPHILE PREAMPS?

As a test, I did something I'd long considered trying but had always put off: I connected my exaSound e38 DAC directly to my power amps. This limits my choice of source components to my server, and to controlling the volume with only the up/down buttons on the exaSound's front panel, or an elusive little Apple remote control. What I discovered was that the effect of inserting each SPL box was almost inaudible. Maybe I perceived a little bit of softening of the highs, but no change in overall balance or resolution. Briefly—though only after a lot of cable-thrashing—I tried the same comparison with the Audio Research MP1 and the Parasound Halo P7 and, dammit, they didn't sound much different either—but both, particularly the ARC, added some high-frequency noise.

That immediately reminded me of the phenomenon of stochastic resonance, which is often cited to explain why one can hear low-level signals that are below the noise threshold. Briefly, the principle is that subthreshold but salient signals, like music, become audible as concurrent frequencies in the random noise add to them.The ear can then distinguish the sum of the signal plus noise as a distinct pattern recognizable as the music signal.

I am positing that the ARC's audible HF noise works the same way to statistically emphasize higher audio frequencies that are below or above threshold. I can hear the hiss from my listening position when I turn the MP1 on, but my ears quickly adapt to it, and since it usually remains on, I'm usually unaware of it. That doesn't mean it isn't contributing a stochastic influence on the audible treble. I take it as consistent with my supposition that the much quieter but still not silent Parasound produced less treble emphasis than the ARC. Running the power amps from the exaSound e38 DAC or via SPL's Volume 8 or SMC 7.1 produced no noise from the speakers unless my ear was actually in contact with a tweeter.

Unfortunately for my hypothesis—from which I am not retreating—the published noise specs for the SPL boxes are not better than that for the Parasound, but they sound quieter than the Parasound or the Audio Research—although the exaSound e38 measures and sounds quietest of all. I just wish it had a volume knob.

—Kalman Rubinson

After a weekend away, I returned and listened again. Now my system sounded really great. Definitely not as it had long sounded, it now had a refreshing new degree of balance. Gone was the occasional high-frequency glint, though that absence was not accompanied by my disappointment over any loss of treble detail; rather, I was pleased with the enhancement of midrange information and its extension into the upper bass. As a result, there were now fewer clues to the physical presence of each of my Bowers & Wilkins 802 D3 front and 804 D3 surround speakers. The entire soundstage was more continuous than contiguous, and the sweet spot was much bigger.

I sat a friend in that sweet spot and played him the highly immersive of Willie Nelson's *Night and Day* (DVD-Audio, Surrounded-By Entertainment SBE-1001-9). He was impressed, as I'd expected him to be. But sitting on another sofa, against the room's the right sidewall, I was stunned to perceive a positive image of the piano directly across the room, against the opposite sidewall, halfway between the left front B&W 802 D3 and the left rear B&W 804 D3. When I described the piano's position to my friend, he replied, "Yes, I hear it there, too!" I find it almost incredible that two listeners, sitting 7' from each other and facing in directions 90° apart, could hear the same soundstage from a system that had been balanced and optimized for only *one* of those seats.

This was an almost freaky re-creation of the continuousness I'd heard from Tom Caulfield's system, but with a bit more breathing room, and enough space to share with friends. In fact, when I played Caulfield's tantalizing preview of an upcoming Channel Classics release—the final song of Mahler's *Das Lied von der Erde*, again with Fischer/Budapest—I felt completely immersed in the venue and the aura: orchestra and singer were in front of me, about 12' beyond my front speakers and extending into the distance. What struck me about what I heard was the lack of what in audiophile circles can masquerade as "air"—a subtle emphasis of higher-frequency noise from the recording venue that forces awareness of its acoustics. Based my attendance at several recording sessions, these sites are generally very quiet. It is at the lower frequencies, where random events energize modal activity, that one "feels" the space. That's what I heard between the notes here.

I couldn't choose between the Volume 8 and the SMC 7.1 based on sound, because to me they sounded identical. I'd opt for the SMC 7.1, if only because it has an additional input for a second multichannel source (*eg*, an Oppo disc player), a pair of stereo sources, and a volume-controlled headphone output. The studio-style switching options are a bonus, though I'd rather have channel-level controls.

Are the SPL boxes completely transparent? Almost—there was a very slight dimming of above 10kHz, in comparison to running the DAC output directly to the power amps. (See sidebar, "Did I Really Prefer SPL's Pro Devices to My Audiophile Preamps?") On the other hand, that seemed ideal with my B&W speaker array. Of course, Dr. Floyd Toole tells us that, trapped in a "circle of confusion," we can't trace a useful reference that will allow us to compare what we hear at home to the original sound of the performance. What I can say is that with the SPL boxes, my system sounds more like Tom Caulfield's—but with better seating.

JRIVER MEDIA CENTER GOES 64-BIT

JRiver Media Center is updated almost continually. The current edition, version 23, was released in mid-June 2017, and patches and enhancements are slipped into it every few days. Now, in mid-October, I'm using build 70! Of course, if your setup is working to your satisfaction, there's no need for you to download every new build; but you can choose to have any new builds (Stable, Latest, or Beta) automatically installed, or you can disable Automatic Updates. I let JRiver automatically install Stable updates, but as a member of JRiver's Beta panel, I do monitor the latest changes because I like to put in my 2¢.

Until September, all Windows, Mac, and Linux versions of JRMC were 32-bit, and 32-bit programs implicitly run less efficiently and use less RAM than their 64-bit counterparts, even on the 64-bit platforms that are ubiquitous today. I've always been impressed by JMRC's speed of operation, but clearly, running it as a 64-bit program should make it better, faster, stronger.

I'm happy to say that the 64-bit Windows version of JRiver Media Center 23 is all of that. Since the 32-bit versions are still being kept current, build numbers for them and for the 64-bit version are the same—you can compare them, if you choose. I didn't bother doing that except to run Benchmark Test, a server system performance test built into JRMC. With the 32-bit version, my benchmark result was a bit over 5000; the same test with the 64-bit version yielded a result of just over 6000—both numbers representing an index that takes into account PCU performance, RAM, disc access, and other pertinent factors. (Test runs vary by ±100.) If you're interested in horsepower, that's an improvement in performance of a healthy 20%.

Of course, the only performance that counts is what it sounds like. Not everyone is torturing their music players by downsampling DSD256 multichannel to 24-bit/192kHz PCM and applying room correction. Even I don't do that *all* the time. Nonetheless, the hardware (and the software it runs) should be able to accept and process current formats without indigestion, even when the diet is rich. With 32-bit JRMC, files of 8GB or more in multichannel DXD seemed to be more of a challenge than DSD256 files, and could result in interruptions for rebuffering—the maximum amount of data that can be loaded into memory for playback is 2GB. Heck, I have 511 tracks that exceed 2GB, and I'll bet many readers of this column have as many. But the cap for 64-bit JRMC is 16GB, and *none* of my files exceeds that.

Since installing the 64-bit version of JRiver Media Center 23, I have heard not a single burp. And it's free—licensees of the 32-bit version can download the 64-bit version and transfer their license. That's a good deal. ■

*Kalman Rubinson (*Stletters@enthusiastnetwork.com*) enjoys being immersed in his music in his Manhattan and Connecticut homes.*

CONTACTS

SPL electronics GmbH
Sohlweg 80
41372 Niederkruechten
Germany
Tel: (49) 2163-98340
Fax: (49) 2163-983420
Web: https://spl.info.en
US distributor: Audio Plus Services
Tel: (800) 663-9352
Fax: (866) 656-0686)
Web: www.audioplusservices.com