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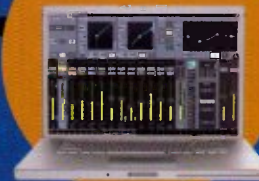
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From the Editor

IMMERSIVE SOUND IN ALL FORMS

For the past few months at *Mix*, we have been completely immersed in Immersive Sound. If you haven't been out to a high-end, refurbished theater lately to see a big film, Immersive Sound is the blanket term for "surround on steroids," a more spherical approach to mix and playback that includes height channels, five across the screen, ceiling speakers, object-based audio, zones and beds and metadata, and seemingly unlimited creative possibilities. While the competing formats from Dolby (Atmos) and Auro Technologies (Auro-3D, licensed to Barco) are vying for position, recently joined by the DTS Multi-Dimensional Audio interoperable file format, plenty of people behind the scenes (including SMPTE Working Groups) are doing their best to avoid a format war that forces theater owners and post facilities to make a choice one way or the other.

In this issue, we asked longtime contributor Larry Blake, who was a *Mix* film sound columnist for more than a decade, to talk to the companies and break down the technologies for our readers. Larry, a re-recording mixer based out of New Orleans best known for his association with Steven Soderbergh, is a natural-born skeptic and is not fearful of expressing his opinion. To accompany his thorough analysis of the formats, he offers an op-ed piece of sorts, where he clearly states the reasons for his built-in bias. It's a good read, whether you work in film sound or not.

We also take a look at the A-list scoring engineer/mixer Dennis Sands' private Santa Barbara facility, which he completely refurbished to accommodate Atmos mixes. While he may seem ahead of his time, Sands has always been that way. And he has his reasons. Give it a read.

Finally, pictured on this month's cover is the Holden Theatre at Sony Pictures Post Production. Originally built in 1996 as part of the 10-stage creative campus in Culver City, the Holden was down for three months in late 2013 while being re-outfitted for both Atmos and Auro-3D mixes, a rare hybrid approach. The first immersive film out of the stage was *The Amazing Spiderman 2* in the spring of 2014.
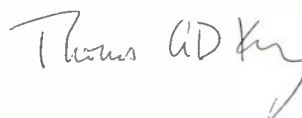
Sony Pictures Post Production has actually been the bridge that has taken our discussions on Immersive Sound from the pages you hold in your hand to a special one-day event—Immersive Sound: From Production to Playback—that *Mix* will be presenting on September 6 at, you guessed it, Sony's studios on the lot in Culver City.

Sony is the host sponsor and has graciously opened its doors to the expected 400-500 attendees. Event partners the Motion Picture Sound Editors guild and the Cinema Audio Society have encouraged their members to attend, and we have put together a comprehensive list of panelists and panels, focusing on both the creative and technical aspects of Immersive Sound.

The main three players—Dolby, Auro Technologies and DTS—along with Yamaha (Nuage) have all signed up as premium sponsors and will be hosting workshops and demos all day in private re-recording stages adjacent to the main editorial panels. There will also be playback demonstrations throughout the day in the facility featured on this month's cover, the William Holden Theatre. Other sponsors include Avid, Meyer Sound, Harrison, RSPE Audio, Fairlight, GC Pro, Vintage King, Imax, Formosa Group and Picture Head.

Immersive Sound: From Production to Playback promises to be a one-of-a-kind event. While the technology is still in its infancy, its acceptance has been accelerating worldwide, among both creative and exhibitors. Come join us as we dive into the biggest advance in film sound of the last 25 years.

Tom Kenny

Tom Kenny

Radial Acquires Jensen Transformers

In summer 2014, Radial Engineering Ltd. in Vancouver, British Columbia, acquired Jensen Transformers Inc., of Chatsworth, Calif. Jensen is now part of the Radial group of companies along with Reamp, Tonebone, Primacoustic and Hafler. Jensen will remain in its Chatsworth facility with an extended lease, an increased inventory of raw parts and newly streamlined manufacturing processes while adding staff to increase its production capacity. In addition, Radial will support Jensen's sales and marketing efforts.

According to Radial President Peter Janis, "Radial will continue to purchase transformers [from Jensen] as it has in the past, as an OEM customer, and Jensen will continue to work with its growing customer base both on the OEM side and as a solution provider for the A/V contractor, home theater installer and professional audio engineer. When Radial was founded in 1992, we started life as a distributor. One of our first product lines was Jensen."

Deane Jensen founded Jensen Transformers in 1974, and Bill Whitlock acquired the company upon Jensen's untimely death in 1989. "After 25 years and my recent turning 70, I felt the time had come for Jensen to be passed along to a younger generation," Whitlock says. "I felt that Peter would be the ideal candidate to take the company on and build it for the future. Peter's company Radial has been tied at the hip with Jensen since 1992 and has been the biggest user of Jensen transformers for years. It only made sense that he takes the reins. I plan to stay on with Jensen assisting on the technical side and working in R&D to develop new and exciting products. There are so many areas that we have yet to explore and with the business side now off my plate, I look forward to focusing on my true love of engineering." ∎

Formosa Group Re-Opens POP Sound Facility

Photo courtesy of The Formosa Group

The former POP Sound facility in Santa Monica, Calif.

In August, Formosa Group, a full service post-production sound company that Picture Head launched in summer 2013, announced that it would reopen the facility formerly known as POP Sound (Pacific Ocean Post) in Santa Monica. Formosa Group is based in historic and state-of-the-art sound facilities on The Lot in West Hollywood, with additional facilities in West Hollywood and West Los Angeles, and is home to respected Supervising Sound Editors, Sound Designers and Re-Recording Mixers.

The POP Sound facility had shuttered on July 7, 2014. Located in the heart of Santa Monica, it comprises 10 mixing stages, eight voice-over booths and an ADR stage. Formosa Group announced that it would refurbish and rebrand the former POP Sound facility.

"We are pleased to re-open this outstanding post-production sound facility," says Robert C. Rosenthal, President and Chief Operating Officer of Formosa Group. "With this addition, we will now have an expanded Westside presence. It also enables the core clientele to continue their great work here with many of the accomplished creative artists and support staff formerly associated with POP." ∎

2014 Emmy Awards for Sound Editing, Sound Mixing

Photo courtesy of The Recording Academy/WireImage

On Saturday, August 16, 2014, the Television Academy presented the 2014 Creative Arts Emmy Awards for programs and individual achievements at the Nokia Theatre L.A. LIVE in Los Angeles. This first ceremony of the 66th Emmy Awards honored guest performers on television dramas and comedy series, as well as the technical disciplines essential to television production. Mix magazine recognizes and congratulates all of the individual winners in three categories of Sound Editing and five categories of Sound Mixing. For the complete list of individual winners, go to mixonline.com.

Pictured here are members of the audio team for the live broadcast of the 56th Grammy Awards broadcast on CBS, several of whom were named as Emmy winners in the Outstanding Sound Mixing For A Variety Series Or Special category. Pictured top row, from left: Mikael Stewart, Andres Arango, Jeff Peterson. Second row from top: Doug Mountain, P&E Wing Senior Executive Director Maureen Droney. Third row from top: Leslie Ann Jones, Glenn Lorbecki. Second row from bottom: BJ Ramone, Hank Neuberger. Bottom row: Eric Schilling, John Harris, Ron Reaves. ∎

DiGiCo, Allen & Heath, Calrec Form New Group
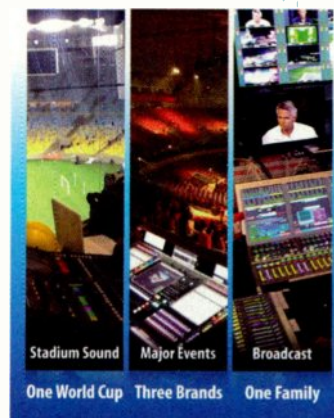
On August 1, DiGiCo announced that it has combined with two other British console manufacturers, Allen & Heath and Calrec, to create a new professional audio group. Within the group, each company retains its particular product specialties, customer relationships and brand identities. James Gordon is Group CEO, supported by managers from across the new group.

"The strategy is to share technology and resource across the group and allow some interconnectivity across the product lines," Gordon says. "We have bold plans for the group, but it is imperative that each company maintains its own independence and style."

This combination required the support of two of London's top private equity houses, Electra Partners and ISIS. "DiGiCo augments our existing investment in the professional audio sector," says Charles Elkington, investment partner at Electra Partners. "We have worked hard over the last 18 months to bring together these three successful businesses."

On the Cover

By Tom Kenny

WILLIAM HOLDEN THEATRE

Sony Pictures Post Production, Culver City, Calif.
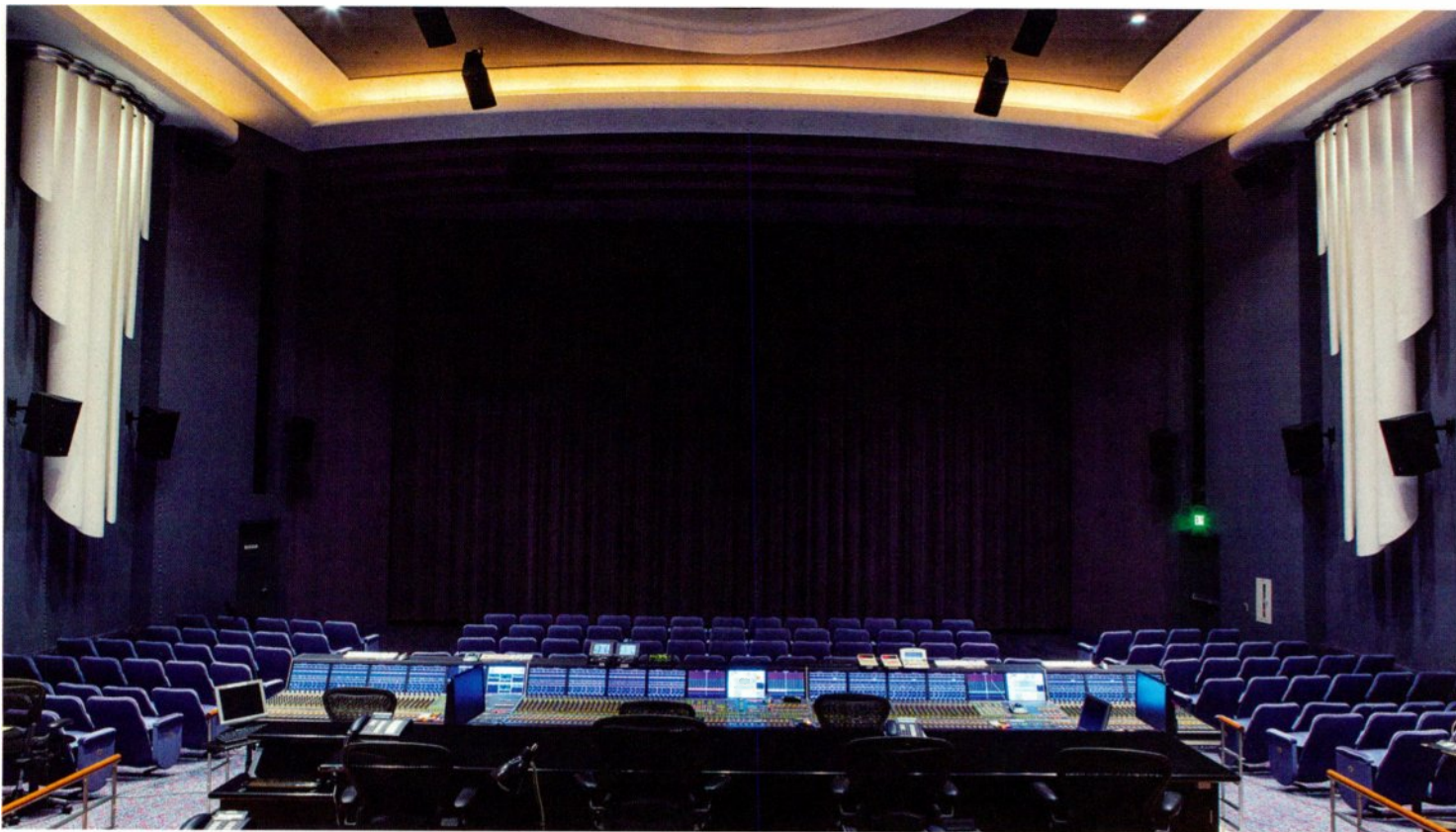
Photo: Matt Dames/Sony Pictures

The emergence of Immersive Sound formats in the past few years represents an enormous leap in motion picture playback, one not seen since the early 1990s and the advent of discrete surrounds and 5.1 digital sound. In many ways it's an even bigger leap, demanding a whole new way of thinking about audio files and placement of individual sounds in a space. Dolby Atmos, Auro-3D and now the interoperable DTS MDA file format, offer the promise of tremendous creative opportunities to sound designers and mixers, at the same time placing new demands and challenges on facilities and workflow. While the competing companies may release monthly updates on the number of re-recording stages and exhibitor screens adopting their particular format, it is important to remember that this is a technology still in its infancy.

Sony Pictures Post Production was by no means the first facility to change over or build a studio for the new formats; in fact, they freely admit that they jumped late. Nor does the company have the greatest number of immersive stages. But when they renovated the famed William Holden Theatre on the Culver City lot beginning in late 2013, they opted for a rather unique hybrid approach, one incorporating both Dolby Atmos and Auro Technologies. The room, which was reworked over a three-month period, came online on January 2, 2014, in time for the mix of *The Amazing Spiderman 2*. It is pictured on this month's cover.

"From a creative standpoint, there was such an excitement when these formats came out—any sound professional is looking for any way to make their work another step better," says Tom McCarthy, executive vice president, post production facilities, Sony Pictures Studios. "So from a creative standpoint, it was easy to make this decision. From a corporate side, when you're looking at making a decision like this, you have to look at the cost. You want to look at any technologies that come out and evaluate and consider them and trust that it's a wise investment, that this investment has legs, that there will be a payback to it. Will there be growth within the technologies that will enhance

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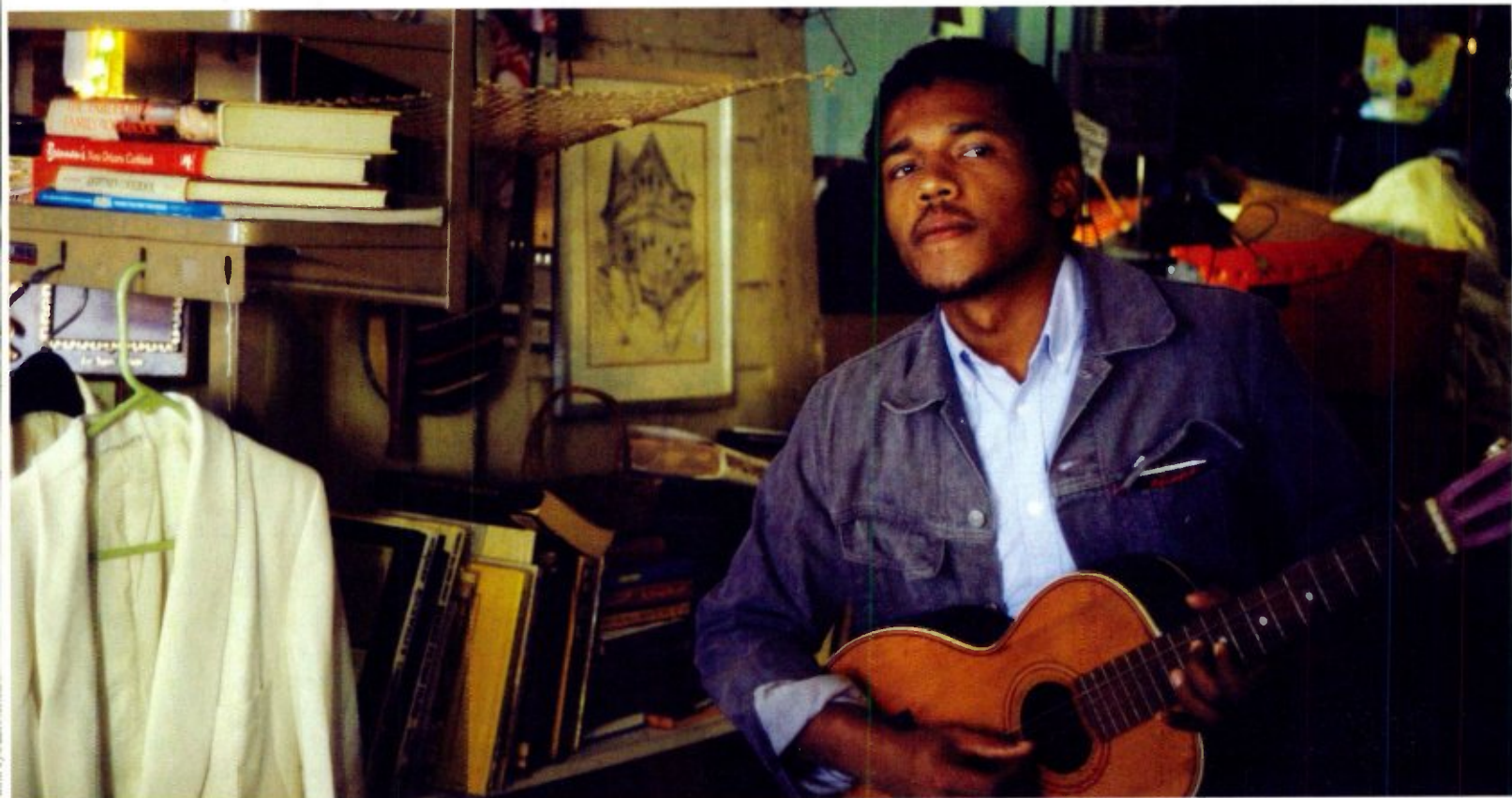
BENJAMIN BOOKER

Explosive Debut for a Newcomer With an Old Soul   By Barbara Schultz

Blues rocker Benjamin Booker is on the road spreading the word about his eponymous debut—sometimes playing his own club dates, other times serving as Jack White's opening act, even joining a few summer festivals, as well. After his appearance at last month's Lollapalooza in Chicago (where a bluesman is certainly preaching to the choir), *Rolling Stone*'s festival review said: "The young, New Orleans-based singer and guitar slinger has found a curiously cool cross between indie-rock distortion, grunge mumble and classic blues-rock that is best experienced live and turned up to II."

Well, Booker's album is pretty great, too. His unusual gravelly, emotional voice has already been compared to Howlin' Wolf's because of its strange beauty (or beautiful strangeness). And this artist's guitar work is riveting—dynamic, distorted, masterful.

Booker made his album in engineer/producer Andrija Tokic's all-analog Bomb Shelter (East Nashville), where the young artist arrived last winter, bringing along just some of his well-written tunes and his only bandmate, drummer Max Norton.

"It was just the two of them at that point, and the studio is still a relatively new experience for them, so I didn't want to overwhelm them with adjustments," Tokic says. "So we kept it safe at the beginning. We just set up vocals, guitar, and drums and started there, recording just the two of them live."

Tokic, whose recent credits also include Hurray for the Riff Raff and Alabama Shakes, set up Norton's kit and Booker's amps in his live room, and put Booker in a booth with sightlines into the main studio. Artist and producer experimented with various amps and instruments to

Andrija Tokic

suit the sound of each song. The main two guitars were a '60s hollow-body Super Lynx Deluxe Vox, and an '80s Ibanez Matsumoku Roadster.

"A buddy of mine runs a company called Philltone in Baltimore," Tokic says. "He has a thing for making cheap guitars really cool, and he put in high-end wiring, replaced the bridge with graphite parts. It has these old Strat single-coil pickups and a '70s Schaller Humbucker in it; it just sounds as ratty as it gets.

"We also used my Harmony amp and a Danelectro Calendar, which is basically a 15-inch speaker and a whole bunch of tubes," Tokic continues. "They're good amps for rock 'n' roll because they're refined enough to get a really solid tone, but if you push them, they break up with a lot of aggression. I think there was a Bassman 100 at one point, too; I disconnected a couple of those power tubes so it was easier to get it to break up."

Tracking to his MCI JH16 2-inch machine. Tokic placed one RCA overhead on Norton's kit, a Neumann U 87 on kick, and a Shure 57 on snare. Booker's vocal chain usually included a Neumann U 67 with an Altec tube preamp and a UA 175 compressor, but nothing was set in stone.

"At one point, I had him just sitting on the floor with a Sennheiser 441 a few feet away from him, picking up his voice and guitar on one channel," Tokic says. "That was for an acoustic thing. Or sometimes I would pair the U 67 with a dynamic mic and blend them together. With Benjamin, separating his guitar from his singing was a weird concept, so we did keep a good bit of the live vocals and reinforced them with overdubs."

With the "band" tracks recorded, Tokic started bringing in bass players (Jem Cohen, Eduardo DuQuesne, Ben Trimble) and keyboardists (Mitch Jones, Peter Keys) to flesh out the recordings. "All the guys we used play different basses," Tokic says. "Jem Cohen plays a cool teardrop hollow-body Vox bass—it's got a really wild old tone to it. Ben Trimble plays a regular Fender Jazz bass. On some songs, you want the bass to be more straightforward—not to grab so much attention to it. But being able to switch to the Vox on songs where we wanted the bass to be really plucky and stand out was great, too.

"It was the same with the organs," he continues. "I have a Hammond A from the late 1930s, and an A100 from the late '50s, and the two guys I had, I knew had unique voices that would serve the songs well. In Nashville, there's just a sea of incredible musicians, and there's a like-mindedness that you find when you're recording exclusively analog. You tend to meet the same people and get a similar vibe."

Tokic mixed Benjamin Booker on his MCI JH600 console. When he says his studio is all analog, he means it. "We don't do anything digital here at all," he says. "I have an EMT plate reverb, a Lawson plate, AKG springs, Orban springs, two different Space Echoes—an echo and a cho- rus echo. Some of these songs are very traditional rock 'n' roll, and it was just a matter of getting the right kind of slap and 'verb, and riding your levels to get the song to stand up enough to keep that performance feel alive.

"But on others, we did all kinds of heavy effecting. We had one track where the song ends and then it comes back as a huge psychedelic reprise. We literally printed half the song to half-inch, and then made a submix mono to a tape track, then remixed all the channels. We remixed the whole outro of the song through a flanging delay and brought up the original submix and the entire new mix, and then blended them together and panned them hard left and right. That was a pretty heavy move!

"One thing that helped through the entire process was, because the experience was so new to him, Benjamin was really open-minded and flexible about trying anything I suggested," Tokic says. "He might have no idea of what I'm doing technically, but he knew a lot of the other music that came out of this studio, and there was a lot of trusting on his part that what he was going to hear back would be rad." ■

Photo by Joshua Shoemaker

Dom Flemons and Guy Davis

DOM FLEMONS, AFTER THE CHOCOLATE DROPS

A founding member of the Carolina Chocolate Drops, Dom Flemons plays guitar, banjo, jug, and an assortment of reed and percussion instruments, in old-time string-band style. He also plays blues and jazz, and his first post-Chocolate Drops solo effort, *Prospect Hill*, brings those American styles together via Flemons' original songs and a varied group of players.

"We got a lot of it live," says Jason Richmond, who recorded the album in Sound Pure Studios and mixed in his personal studio (both in Durham, N.C.). "A lot of songs started with Dom and [guitarist] Guy Davis playing off of each other in the studio, facing each other. We'd start with live sessions, and then build tracks up with overdubs."

Flemons does almost all of the singing, and Richmond captured those parts to Pro Tools with a Brauner VMX microphone through a Chandler TG2 pre and an Anthony DeMaria Labs 1500 compressor. "That Brauner mic is mine," Richmond says. "It's not as clean as their VM1, but it's still modern-sounding. It has such a great frequency response, especially for low frequencies, and the top is incredibly smooth. There are a couple of other vocalists on the album, and I ended up using Telefunken mics on them for different color, but the Brauner is an all-around stellar mic."

On Flemons' string instruments, Richmond put up a Coles 4038 ribbon and a Schoeps MK4. "I have a tendency to double-mike things so I can pick and choose later," he says. "With Dom, there's a lot of chasing him around the studio while he tries different things. I would throw up multiple mics so I'll have choices down the road."—*Barbara Schultz*

COOL SPIN
SINEAD O'CONNOR, 'I'M NOT BOSSY, I'M THE BOSS'
(NETTWERK)

Hardly anyone writes about Sinead O'Connor and doesn't mention some personal or political controversy, but her music is so worthy of unqualified praise. The new album, *I'm Not Bossy, I'm the Boss*, is intimate, emotional, sensual—focused on the reverb-laden, chorused vocals that have been the hallmark of O'Connor's best records. She and her producer, John Reynolds, are masters at creating vocal dynamics. Sweet, layered vocal harmonies are undercut by a banshee wail in dark moments, or those harmonies will peel away to reveal her startling, raw unprocessed howl, or a whisper. What's actually shocking about O'Connor is the depth of love and anguish she's able to convey with her wondrous and explosive songs. Open your ears and forget the rest.

Producer/engineer: John Reynolds. Mixed by Reynolds and Tim Oliver. Additional engineering: Graham Bolger. Studio: New Air Studios (London). Mastering: Kevin Metcalfe/Soundmasters (London). —*Barbara Schultz*

A YEAR AT PANORAMIC HOUSE

John Baccigaluppi

Panoramic House is a personal recording studio and vacation home owned by John Baccigaluppi and Bobby Lurie, located on the western Marin County coastline north of San Francisco. The duo, who also own and operate The Dock in Sacramento, Calif., and Mavericks in New York City, renovated the historic 1960s structure, which was built entirely from recycled architectural materials salvaged from the Bay Area. Now, Panoramic House is celebrating its first year of operation with equipment relocated from Baccigaluppi's former recording studio, The Hangar, which includes his recently expanded API 1608.

When the five-year-old 1608 moved to Panoramic House, it was expanded to 32 channels to accommodate the studio's 16- and 24-track analog tape decks. "With more analog tracks, we needed more console real estate," says Baccigaluppi. "The 1608 has served me very well. I love that it has the same API circuitry that runs its large-format consoles. Eight aux sends and eight groups paired with the modular 500-series slots make it a very versatile board. We have a lot of EQ flavors, including a handful of API 550As, 550Bs, and 560s.

"Panoramic House is definitely analog-centric," he continues. "The first console I ever worked on was an API 2488 at The Evergreen State College in Olympia, Wash., and I love that the 1608 continues in that tradition. Moreover, clients are always impressed. API is a trusted name in the industry."

Panoramic House has already been host to several big-name indie bands, including My Morning Jacket, Band of Horses and Thee Oh Sees. "It's funny," Baccigaluppi concludes, "that most of the vacation renters come to town during the summer when it's foggy to the point that you feel like you're in a cloud. Conveniently, most musicians are touring during the summer. The other three seasons are lovely, and that's when most musicians want to settle in and create. Plus, that's when the surfing is the best!"

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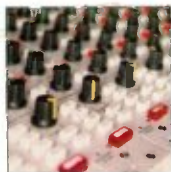
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Double-Dose of Lucinda Williams

'DOWN WHERE THE SPIRIT MEETS THE BONE'

By Barbara Schultz

Lucinda Williams first visited David Bianco's North Hollywood Studio, Dave's Room, in December 2012. Bianco had agreed to donate studio time for a benefit album to assist Replacements guitarist Slim Dunlap, who had suffered a stroke earlier that year. The session was part of New West Records' Songs for Slim project, and featured performances by Jakob Dylan, Steve Earle, Joe Henry and John Doe, as well as Williams, backed by a band of creative and like-minded musicians.

"Lucinda came in for the benefit, and she really took to the studio," recalls Bianco, who has also engineered Number One albums for Bob Dylan, Mick Jagger, Big Head Todd & the Monsters and others. "Her husband, Tom [Overby], pulled me aside and said, 'We've got to do some records, and Lu seems really comfortable here. We'd like to come back.'"

Williams returned the following October to make a double album, *Down Where the Spirit Meets the Bone*, with co-producers Overby and Greg Leisz—who also contributed his guitar

brilliance to the project—and a rhythm section comprising the band Jack Shit (guitarist Val Mc-Callum, drummer Pete Thomas and bassist Davey Faragher). Sessions were wedged in between tour dates, with guest musicians including guitarist Bill Frisell, keyboardist Ian McLagan, Jakob Dylan and more joining on select songs.

"The song 'It's Gonna Rain' was the only song we did with [Williams' longtime] drummer Butch Norton and bass player David Sutton as the rhythm section, with Bill Frisell and Greg on guitars and Jakob doing harmony vocals," Bianco explains. "But there's a whole other album in the can with Butch, David, Greg and Bill that is planned to be released next year."

Whatever the lineup, Bianco captured the group live in Dave's Room, where's he's been an owner/operator since 2006. "My studio used to be called Mama Jo's, and it was purpose-built from the ground up to be a studio around 1970," he says. "Vincent van Haaff designed the control room, and the owner, [producer/engineer] Freddie Piro, wanted him to model it after Lar-rabee Studio B. I had done a lot of work at Lar-

rabee, so when I walked in, it was like déjà vu."

When Bianco moved his Pro Tools HDX system and his collection of outboard gear and mics into the studio, it was four years' empty and the worse for no wear. He recalls finding creative ways to bring the rooms back up to par: "I bartered with studio time," he says. "I recorded a bass player who could lay flooring. I had a plumber-drummer."

Luckily, the excellent bones and lines of the studio were still intact. The recording spaces include a live drum room flanked by four iso rooms of varying acoustics and sizes, which allowed Williams and band to record live, yet keep a pure signal on each piece.

The musicians and producers arranged Williams' poetic Americana tracks in the studio, working off of her homemade demos. "I had Lu in a large vocal booth, and Ian McLagan in the keyboard room, and we had the guitar amps sequestered off in different booths, so most everybody was out in the room playing and had great sightlines to each other."

Bianco captured Williams' sultry voice with a Mojave MA-300 microphone, through a preamp in his Neve Sidecar, into a UREI LA4A limiter. "I have an old EMT 140 plate that Herb Alpert cherry-picked out of A&M Studios before he left," Bianco says. "She loved that reverb and a little bit of slap delay."

Likewise on all of the instruments, Bianco mainly employed his go-to method of pairing carefully chosen mics with vintage (or vintage-style) analog pre's and compression, straight into Pro Tools.

"The idea is to get the best of the analog and digital worlds, so I have my Neve Sidecar and Quad 8 and API preamps; those form the bulk of my warm analog elements, along with some really nice old compressors—1176s and LA-2As, dbx 160s and so forth. But I also have some modern pre's that I like to mix it up with: Great River, Phoenix Audio, SSL and Digi pre's. For example, I use the Neve for nice, warm, big drum sounds, but for the clarity I like for overheads, I would go for a Great River or something else."

Bianco mixed the record in Dave's Room, as well, with final tweak comments from Overby, Leisz and the artist herself. "I'd already provided Lucinda with rough mixes that she really loved," Bianco says. "Most of what made the songs really came from the great interaction between the musicians playing off each other in the studio—the stuff that happens when you get everyone into a room and finally hear everything in 'living color.'"

In 1973, at Watkins Glen, live sound went just a bit digital.

"In the beginning, audio was analog. And, almost anything was possible."

Everything but delay. And so the Digital Delay Line (DDL) was born. DDLs were simple and limited to just delay. Today, forty one years after a stack of Eventide DDLs delayed the audio feeds to the speaker towers at Watkins Glen, Eventide introduces the **DDL·500**. Designed to be 'as analog as possible', the **DDL·500** will make your 500 series lunch box "just a bit digital."

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Classic Tracks

By Blair Jackson
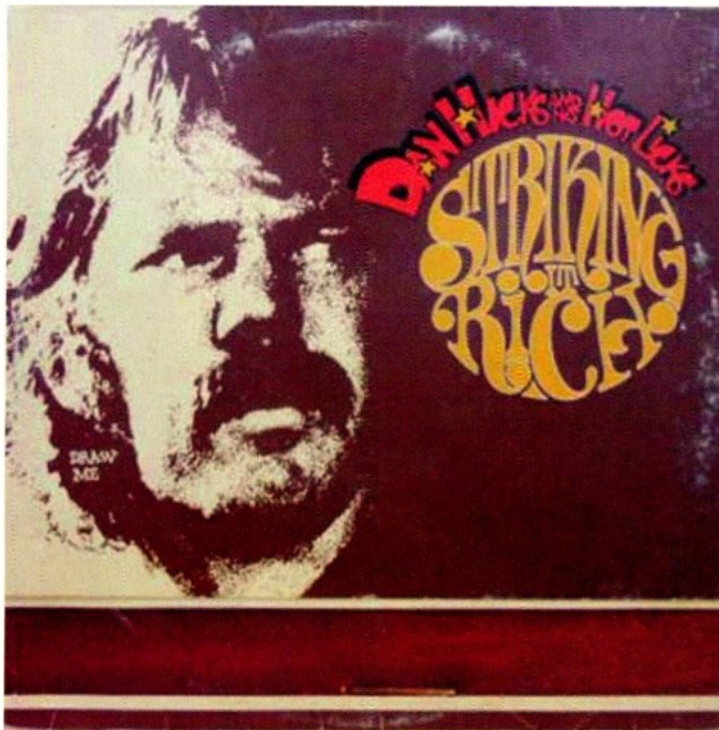
"I SCARE MYSELF"
Dan Hicks & His Hot Licks

One of the most interesting and engaging groups to burst onto the pop music scene in the early 1970s was a San Francisco Bay Area outfit known as Dan Hicks & His Hot Licks, who fashioned an eclectic amalgam of retro music styles into something quite fresh and original. The wry, dry and clever singer-songwriter-guitarist Hicks delivered quirky songs that drew musical inspiration from Big Band vocal tunes, the swinging "hot jazz" of Django Reinhardt and Stéphane Grappelli, cowboy songs, '20s and '30s blues, various Latin styles and American folk sources, from bluegrass to jug bands. Besides leader Hicks, the prime version of the Hot Licks included the spectacular violinist "Symphony" Sid Page, the fleet-fingered guitarist John Girton, stand-up bassist Jaime Leopold and the wonderful harmony singers Maryann Price and Naomi Ruth Eisenberg.

The group emerged at a time when adventurous FM rock radio ruled the roost in many U.S. cities, so Hicks' idiosyncratic, largely acoustic sound provided a welcome counterpoint to (or respite from) the big rock outfits of the day, such as the Allman Brothers, Derek & the Dominos, Yes, Jethro Tull, Elton John, etc. This month's Classic Track is perhaps the group's best-known song, "I Scare Myself," which actually appeared on two different Hot Licks albums—a little-noticed 1969 effort for Epic called *Original Recordings*, featuring a different guitarist and backup singers, and the version we're highlighting, from the group's masterful and critically acclaimed 1972 opus, *Striking It Rich*.

Originally from Santa Rosa, north of San Francisco, Hicks was a drummer in his high school years but took up the guitar in college at San Francisco State during the folk music boom. Still, he was drafted to be the drummer in what turned out to be the first psychedelic band of Haight-Ashbury's counterculture era, The Charlatans, who were as famous for their eccentric Edwardian/Old West look as the rather conventional blues- and folk-rock they played. Never as original or musically proficient as the Grateful Dead, Jefferson Airplane, Quicksilver Messenger Service or any number of other local bands, The Charlatans had a star-crossed few years before dissolving at the end of the '60s. Far from being just the drummer in The Charlatans, Hicks also sang, played guitar and was the group's best songwriter, turning out a number of tunes he would later transform and bring to the Hot Licks, including "How Can I Miss You When You Won't Go Away," "By Hook or By Crook" and "'Long Come a Viper."

By the time The Charlatans breathed their last, Hicks was already playing clubs as a solo artist around San Francisco. "At some point in there," Hicks says, "I made a demo of my songs at Columbus Recorders with Nick Reynolds [of the Kingston Trio, whose manager, Frank Werber, owned the SF studio], and I got some accompaniment for that. I think that's maybe how I started adding people to my act—Jaime Leopold on bass and David LaFlamme on violin." LaFlamme went on to front It's a Beautiful Day (famous for "White Bird"), but Hicks liked having the violin accompaniment to his songs and soon enlisted Sid Page to fill the role.

"After I had the violin and the bass," Hicks recalls, "I got the idea of two girl singers and went through a couple of different pairs. That gave me another reason to compose and create involving the girls with call-and-response or three-part harmony. I got more prolific. I'm pretty sure 'I Scare Myself' came out of that. The songs that turned up on the first Epic record were a product of that new stimulus.

"I'm not exactly sure when I wrote 'I Scare Myself,'" he continues, "but I have a story in my mind of how I started the whole idea of it. I was living in the Haight and I went over to Sausalito to visit Nick and Joan Reynolds—Joan had a club in North Beach I used to play [in the late '60s] called the F.W. Kuh Memorial Auditorium. I wasn't much of a doper or a drinker at that time; that came later. During this period I mostly drank coffee and wrote songs and played gigs. But [the Reynolds] had some kind of hash or marijuana cookie, and I had a bite of that and I remember getting in the car and driving back to the city over the Golden Gate Bridge, and the phrase came to me: 'I scare myself,' because that's how I

Dan Hicks & His Hot Licks in San Diego, 1973. (L to R) John Girton, Maryann Price, Jaime Leopold, Naomi Ruth Eisenberg, Dan Hicks, Sid Page
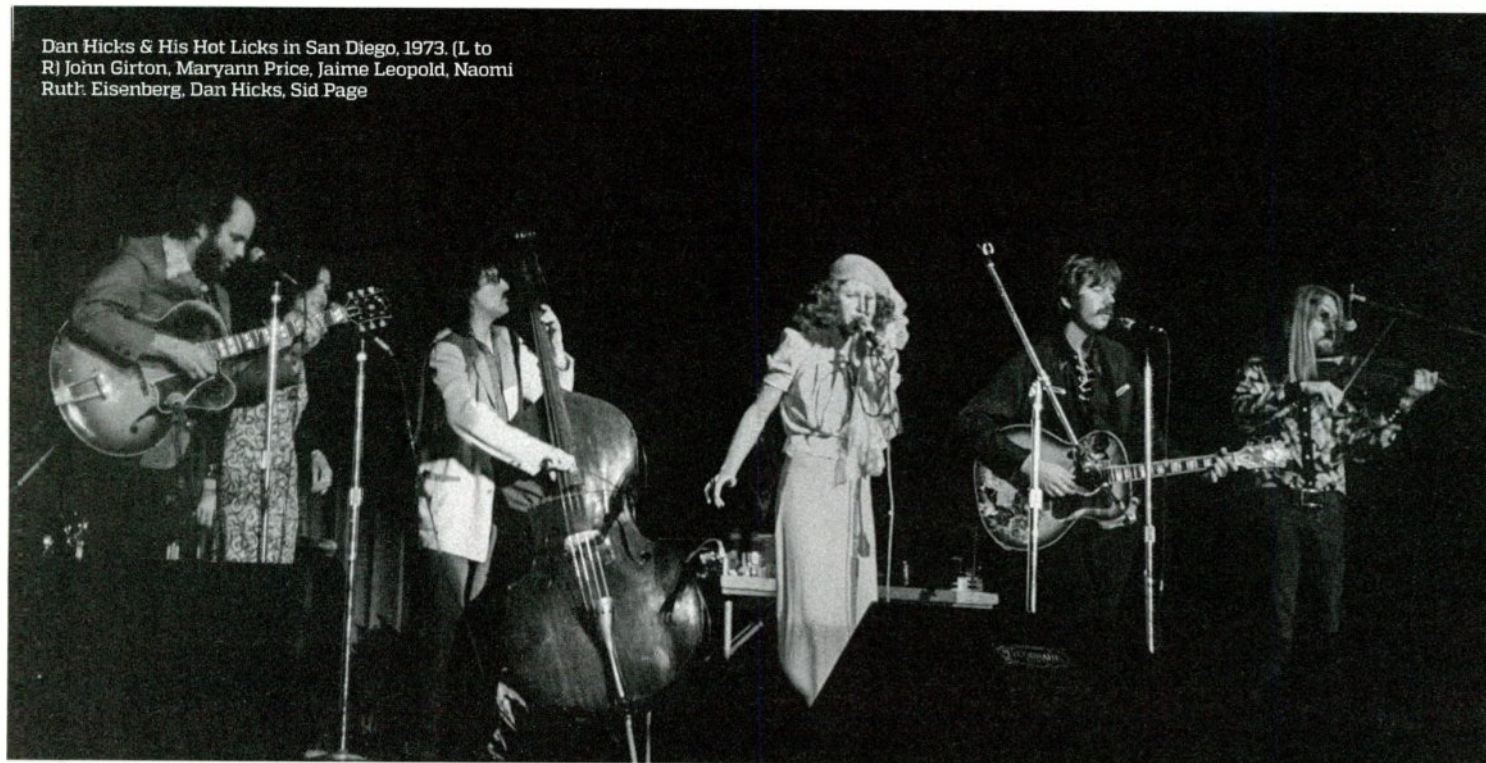
Photo: Dave Collis

was feeling—a little paranoid and uncomfortable. I don't know if I saved that idea, or as soon as I got home started playing guitar. I used to carry around a little notebook and I'd write down titles or things that came to me. I think the whole love song part of it probably came pretty quickly, and I don't have any recollection of it going in any other direction."

As for the Spanish-influenced musical setting for the lyrics, "Maybe it was from me liking [Ravel's] 'Bolero'; I don't know," Hicks offers. "I started out as a drummer and I played a lot of dance gigs and dance classes up in Santa Rosa, so there were different Latin beats. But I didn't want it to be a cut-and-dried Latin song."

The ill-fated Epic album on which the song first appeared was cut at Columbia Studios in L.A. with producer Bob Johnston. By the time the next Dan Hicks & His Hot Licks album came out two years later—a live set called Where's the Money?—the group had a new label (Blue Thumb), Price and Eisenberg comfortable as backup singers (who occasionally sang lead), no guitarist, and a new recording team: producer Tommy LiPuma and engineer Bruce Botnick.

LiPuma, one of the principals in the recently formed Blue Thumb operation, had already enjoyed a substantial career doing A&R and production at Liberty Records and Herb Alpert's A&M label, "but I felt maybe I was being typecast a bit because I'd had hits with the [easy-listening vocal group] The Sandpipers, and then I was asked by Herb to do Claudine Longet [also easy listening], and that was a hit, too. But that's not where I wanted to end up." After seeing Hicks' group play a show in L.A., he signed them to Blue Thumb and then brought in Botnick—with whom he had worked on many projects (though not on the Doors albums that made Botnick famous)—to record a live album at the Troubadour nightclub in L.A. That album drew raves from the rock press, and the group's first national tour, with guitarist Girton now on board, was very well received.

When it came time for the group to record a studio album, it was a no-brainer figuring out where: Sunset Sound in Hollywood had seen many a session with LiPuma and Botnick, together and separately. So

in January of 1972, Hicks and company headed down to L.A. to record Striking It Rich in Studio One at Sunset. Hicks had a bunch of recent tunes planned for the album, and at LiPuma's request, two songs that had originally been recorded for the Epic album were re-cut: "I Scare Myself" and "Canned Music." LiPuma says, "Dan gave me a little bit of a hard time about that—'Man, we've already done those!,' but I managed to convince him: 'Let's try it, and if they don't come off, forget it.' Well, they came off really well."

At that time, Studio One had recently replaced its original Alan Emig-designed tube console with a new solid-state one built by Frank De Medio. "It was a shock going from tube to solid-state, and not an improvement by any means," Botnick recalls. "The sound wasn't as open; it didn't breathe as much. It took awhile to figure that out and how to deal with it. It was almost like going to a new studio, and I'd been there for years." However, the De Medio console had 24 inputs (versus 12 on the tube board), with easy routing and monitoring. "There was a drastic shift in tonality, but it's a matter of understanding what you're listening to and then working around it. We had [Universal Audio] 176s [for compressors] and the EQs were all Pultec, which did mitigate some of the problems with the console. And if I go in there today and mix on that desk, it sounds great; a version of it is still there."

LiPuma elected to record the album completely live, with all the players in the main recording room, eschewing the vocal booth altogether. "What we ended up doing," LiPuma says, "was set them up in a semi-circle, and each of them had a mic, so everyone not only had eye contact, but they had a sense of what each was doing. They had earphones on, in some cases just one, to make sure their pitch was on. But it was totally live, recorded in the room, vocals and all."

In Botnick's recollection, "We may have done it on 8-track; you didn't really need more, because you had the four instruments [two guitars, vi-

CLAIR BROS. INTRODUCES CLAIR SOLUTIONS

CLAIR solutions

On August 1, Clair Bros. Audio Systems Inc. in Manheim, Penn., announced a new trade name and logo for its audio, video and lighting integration division: Clair Solutions, which now serves as the company's sole representation in the A/V/L integration industry. Clair Brothers manufacturing will continue to concentrate on sales and distribution of professional loudspeaker systems both domestically and worldwide.

"We understand there's some confusion and we'd like to clear it up," states founder and CEO, Roy Clair, in a video announcing this new venture. "The mission of Clair Solutions is to provide the right solution for the customer's project, no matter the size of the organization or venue and regardless of the equipment manufacturer."

The company dedicated the name Clair Solutions to its integration services in order to better communicate the core offerings of that division. The previously established Clair Brothers brand will continue to operate as a professional audio products manufacturer. As a result, each division will be able to function independently and focus on separate and distinct identities.

Clair Solutions offers a range of integration services, including audio reinforcement, acoustics, video production, theatrical lighting, staging, rigging and controls systems for performance venues, houses of worship, sports facilities, educational/corporate facilities, hotels, clubs, restaurants and cruise ships. ∎

PASSENGER IN THE U.S.

Photo: Todd Berkowitz

Passenger at Philadelphia's Electric Factory in August 2014.

Singer/songwriter Passenger (aka Mike Rosenberg from Brighton, UK) is touring North America along with Canadian trio The Once and Stu Larsen in support of his latest release, *Whispers*. After this leg concludes on September 14 in San Diego, Passenger will head to Europe in October.

Passenger is touring with front-of-house engineer Simon Kemp and monitor engineer Russ Matterson. The monitor control package is provided by Special Event Services in North Carolina, including SES' custom 12-inch wedges. Kemp says that in the U.S. they are relying on house-provided P.A. systems and that they are carrying a pair of DiGiCo SD9s for monitors and FOH. "We gain share a DiGiCo D-Rack [stage box]," Kemp says. "It works really well. When we play festivals we use two DiGiCo SD11s that we fly with, and our small RF rack."

"The vocal mic of choice is an AKG D7, which has all the characteristics of a vocal condenser but is actually dynamic," Kemp continues. "This show is all about vocal power and intelligibility so I need a lot of gain before feedback. Mike is also completely wireless with his guitars and his stompbox. For this we use AKG DMS 700 digital wireless system. This has worked brilliantly, and after lots of time spent comparing it to a wired guitar method it sounds exactly the same. There is no loss or any of the classic RF sounding issues you would normally get."

Kemp's mix of Passenger is straightforward, with an interesting twist: "I don't use plug-ins," he says. "I'm a great believer in getting it correct at the beginning and keeping it simple. I do a few tricks with his guitar, and use the subs for some real guts. The rest is really just effects and delays. The main weight of the mix comes from his shoe microphone: This is an AKG transducer built into his shoe. The result is a big 808 sounding bass drum." — *Matt Gallagher*

PALAIS EN JAZZ FESTIVAL

Photo: Joëlle Rasoarivelo

The Palais en Jazz event at King Louis XV's Imperial Palace Compiègne in northern France featured Amadeus P.A. speakers and stage monitors.

On June 28 and 29, Palais de Compiègne, a royal palace in northern France built in the 18th century for Louis XV and later the residence of Napoleon I and Napoleon III, provided the setting for the third annual outdoor event Palais en Jazz. This year's festival included composer/pianist Michel Legrand's trio, among others. Sound reinforcement included Amadeus' PMX, DIVA and MAESTRO speaker systems; an SSL Live L500 console; and Sonic Emotion's Wave 1 3D Sound Processor. The three companies were among the event's co-sponsors.

Front-of-house engineer Emmanuel Feyrabend mixed his first Palais en Jazz event this summer. "When I arrived on-site, I was introduced to the WFS [Sonic Emotion's Wave Field Synthesis] process by Arnault Damien from [French distributor] Euphonia. He had been charged by the festival to implement this arrangement in connection with the standard Amadeus-built system designed by Michel Deluc. There was little time to do the Michel Legrand Trio soundcheck. It was pouring rain, and the 'horizontal' system was shielded from the rain. Given the time constraints, I opted for safety and focused on the vertical DIVA XL system.

"After the soundcheck, with engineers Arnault Damien and Lucas Hourdin from Juke Box Limited, we decided to feed the WFS system with post-fader stems from all instruments and voices," Feyrabend continues. "Arnault was able to deliver to each and every listener an identical spatial imaging of the show. The FOH system simply disappeared in terms of where the sound sources originated." ∎

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ARCADE FIRE

Win Butler

Regine Chassagne

Mix caught Arcade Fire—Win Butler, Regine Chassagne, Will Butler, Richard Reed Parry, Tim Kingsbury, Jeremy Gara and Sarah Neufeld—last month when they performed at The Forum in Los Angeles.

"I've been using the Avid D-Show for nearly 10 years now," says **FOH engineer Jim Warren**. "I warned the band at the very start of the tour that if they wanted to go over 96 inputs, I would have to find a different console to use, and they insisted we'd never get to that number of inputs. We got there in the second week of rehearsals! In spite of being worked hard, it has performed flawlessly throughout the year. The only outboard gear I use are a Moogerfooger and Space Echo—two delay effects pedals that sit up against the console meter bridge," he continues. "Everything else the console's onboard EQ and dynamics processing, or plug-ins.

"The band's arrangements tend to be pretty dense, so the first task was to achieve as much separation as possible between instruments on a song by song basis," he continues. "The majority of the inputs have the EQ and pan recalled on a snapshot, which helps with fine-tuning and creating space. The second step is to mix the show very dynamically and ride in and out all the 'hooks.' It is impossible to have everything louder than everything else. I would definitely say I am aiming at a 'live' mix. The comment I always hear about Arcade Fire live shows is the incredible energy that they generate onstage, and I think when you're trying to convey that energy, it's important to allow the mix to breathe and to give the impression that everything's a little out of control. In reality, of course, while allowing one thing to go crazy—quite often with Will Butler on the other end of it—I'm reining in other stuff to make room for it."

"I am mixing on a DiGiCo SD7," says **monitor engineer Jon "JonO" Ormesher**. "There are at least 12 people onstage, so I need a board that will accommodate my 52 outputs. I am using internal FX/Gates/Comps—no need for any outboard equipment.

"For wedges I am using a combination of d&b audiotechnik M2s and M4s, along with a ButtKicker for the drummer and a sub for one of the keyboard players," he continues. "The percussion players are on wedges alone and the brass are on ears; everyone else has wedges and ears. With this band, it's not one mix position per band member, as they all swap instruments frequently, sometimes during a song. We have 12 vocal mic positions on stage, and these have to be covered for all eventualities. We are using Sennheiser RF units and a combination of UE and Sensaphonic IEMs.

"My monitor tech is Yvan Sauve, who is an immense help in not only setting up the stage but in keeping me relatively sane. He has a very busy workload. There are 92 inputs to plug in and he does it with aplomb every show. I have to mention Mathieu Levesque for his tireless work on our RF. We have at least 16 RF inputs and 18 RF IEMs, which makes for a busy time these days, especially in Europe where the RF 'rules' are changing in every country. They are both a credit to Solotech."

Will Butler

Richard Reed Parry

Tim Kingsbury

Sarah Neufeld

Solotech of Montréal is the P.A. provider, with a main hang of 14 L-Acoustics K1s with three to six KARA downfills per side, depending on the venue. Eight K1 SB subs are flown per side, beside the K1, with an additional 14 K2 per side. There are also eight SB 28s per side on the floor, and six L-Acoustics KARAs as front fill. The B-stage, flown in front of the FOH position, is a full circle of 16 L-Acoustics ARCS and subs.

"I'm carrying a Meyer Sound SIM3 analyzer and two Meyer Galileos for all the EQ and time alignment," says **systems engineer Marc-Oliver "Marco" Germain**. "Signal distribution is all done through Riedel Rocknet, AES/EBU with analog redundancy."
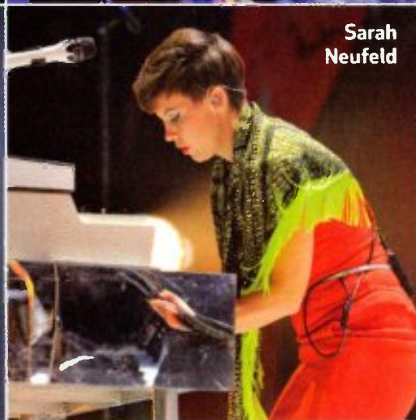
The audio techs, from left: Mathieu Levesque (wireless and stage tech); Marc-André Théoret (P.A. tech); Yvan Sauvé (monitor tech).
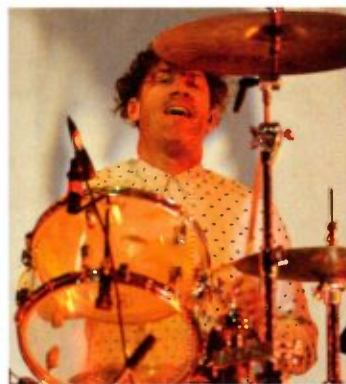
Microphones on drummer Jeremy Gara's kit are: Shure SM91 (inside kick), Beyer M88 (outside); Sennheiser 604 (snare top), Shure Beta 98 (snare bottom), Beyer 201 (side); hi-hats (RØDE M5); Sennheiser 604 (toms); Audio-Technica AT4050 (kit ambience and overheads).

Immersive Sound for Cinema

By Larry Blake

The introduction in 2005 of the Digital Cinema Initiatives standard brought with it the largest wholesale change in motion picture presentation since the arrival of widescreen cinema and stereophonic sound in 1953. It differed greatly from the past because picture and sound specifications had already been carefully vetted by committees with an eye toward scalability of the DCPs (Digital Cinema Packages) that are sent to theaters.

For the image, this meant 2k resolution was the minimum, but 4k was supported; in sound, all theaters were expected to have basic 5.1 systems, although the standard allowed for a total of 14 channels. Two additional channels are reserved for mono mixes for hearing impaired and visually impaired patrons, the latter being narration on top of the mix.

However, it was inevitable that variations would soon occur, and these were first in picture with various implementations of 3-D. As soon as this was starting to sort itself out in 2012, two different immersive sound formats arrived to break the 7.1 barrier that was the limit for almost all previous DCPs.

First, in January 2012 Auro Technologies, in association with Barco Cinema, introduced Auro-3D with the film Red Tails in Auro 11.1, which was shown in about 2 theaters in the U.S. The development of Auro-3D began seven years prior, with research that CEO Wilfried Van Baelen had done at his Galaxy Studios in Belgium.

The Auro-3D cinema format, in its basic 11.1 cinema iteration, adds a 5.0 height layer—three screen speakers and two upper surround channels—above the standard 5.1 system—plus a top layer comprising a center-ceiling "Voice of God" channel. The system can be expanded to 13.1 with the splitting of the lower surrounds into four channels, as in 7.1.

Utilizing their proprietary Auro Codec, the additional tracks are encoded in the four least significant bits of a standard 24-bit, 48kHz mix, so that only one 5.1 or 7.1 printmaster needs to be shipped on DCPs, with the additional height and top channels decoded in the cinema.

Auro Technologies has a complete suite of plug-ins to aid mixers, including the Auro-Panner, to place sounds in the 3-D field, and Auro-Matic Pro, which allows upmixing of mono, stereo and 5.1 elements to their 11.1 and 13.1 formats.

The second "salvo" in the new format wars occurred in June 2012 when Dolby Laboratories introduced its Atmos format for the Pixar animated film Brave on 14 screens. Dolby had been researching expanding cinema speakers

for years, going back to 2002 and We Were Soldiers, which utilized an overhead VOG channel.

After years of experimentation with various speaker positions, including screen height as in Auro-3D, Dolby arrived at standards for surround speaker spacing, locations, and dispersion and mounting angles. The side surround speakers begin near the screen, and fill up the first third of the auditorium where normal surrounds are absent.

Timbre match of surrounds to screen channels is made a reality by employing bass management; this, combined with the placement of surrounds closer to the screen, helps smooth out the transition of sounds off the screen and by giving surrounds much increased power handling. Bass management is not used in all theaters; at Dolby's screening room in Los Angeles and at the Samuel Goldwyn Theater at the Academy, the existing surrounds were able to go down to 40 Hz, which matches the specified low-end response of screen speakers. The final speakers added in Atmos are two overhead arrays down the length of the theater's ceiling. Up to 64 speakers are supported by the CP-850 Atmos cinema processor, which went into production in April 2013; before that, theaters were using the studio RMU mastering unit.

OBJECT-BASED AUDIO

Where Auro-3D is its current form is channel-based in the classic stereo film manner, with recorded tracks assigned either to specific speakers or arrays of speakers, Atmos is object-based. In object-based cinema audio, sounds are not necessarily dedicated to a specific channels for the length of program, but instead individual files are placed in the three-dimensional space of the theater via metadata containing level, location XYZ coordinates and start/stop times. (X is left-right across the screen, Y is from the screen to the back wall, and Z is height.)

Object-based audio (OBA) is of course the foundation of video games, in which the timing and location of sounds are variable according to where players are in their worlds. For movies, which occur in a linear fashion, OBA is used for two purposes: One, to pinpoint the location of a sound in what otherwise might have been an array (such as a surround theater wall) or a group of speakers (such as behind the screen) or in three-dimensional variations among arrays and speakers. Two, it allows for this accurate panning to take place in various theater configurations and sizes: "halfway down the right side wall" scales to the same position, regardless of whether the wall contains eight or four speakers.

Among the first public demonstrations of OBA for cinema were in the early part of the last decade by IOSONO, based on research done at the Fraunhofer Institute in Germany. IOSONO was shown in various venues in Los Angeles from 2008-2010, although current IOSONO efforts have primarily been in special venues and corporate events. As of summer 2014 the company is undergoing financial restructuring.

While it is possible to mix Atmos exclusively utilizing objects, standard practice has evolved that mixers have "beds," which are essentially full-length "static object" tracks dedicated to specific channels. Thus, 7.1 beds for the dialog, music and effects stems at the final mix involve dedicating 24 of the 128 inputs to Dolby's RMU. Sound effects and music beds are frequently expanded to 9.1, with the stereo overheads as two arrays.

In this example, up to 104 object tracks can be recorded as mono .wav files containing XYZ coordinates and other object metadata. While most mixes may never need 128 simultaneous objects, object tracks (like beds) are dedicated to specific stems, to easily allow the creation of M&Es, not to mention facilitating archiving. When creating the MXF-wrapped file that is in effect the printmaster of Atmos mixes, only the actual audio used in the mix is used, with the silence between the events on all tracks deleted—objects or beds—are deleted for space-saving purposes. During rendering in the theater, the objects are triggered in sync and placed to the proper locations according to the metadata.

The "scalability" of Dolby's Atmos specifications apply not only to the surround and overhead arrays, but also to the screen, specifically the left-center and right-center speakers that have recently been largely absent from film mixes, save for certain Sony Dynamic Digital Sound (SDDS) mixes that used all channels in that format. (The configuration of course began in the Fifties with Cinerama, later continued in Todd-AO.)

Dolby has been strongly recommending Lc and Rc speakers where the screen is wider than 40 feet, and reports that a large percentage of Atmos installations have five screen channels. The presence of those speakers will make themselves known by smoothing out pans in wide screens common in today's cinemas with stadium seating, especially to patrons seated close.

While the smoothing of pans with standard three-screen speaker mixes, further benefit can be had when mixers create stems with five screen channels or create "static objects," assigning elements to the narrow-width Lc and Rc speakers. This is generally regarded as very useful for dialog and effects panning, and for increasing the resolution of the primary screen "proscenium." Again, just as three-speaker mixes spread out naturally to five, phantom images are created for Lc and Rc objects when there are no speakers present.

INDUSTRY STANDARDIZATION

When digital sound first came to film exhibition in the early 1990s, three formats competed for the attention of filmmakers and theater owners: Dolby Digital, DTS and SDDS. Initially, distributors were divided into "camps," so to hear any film in digital stereo, exhibitors had to install all three formats, something that was not practical or cost-efficient.

By about 1995, theater owners were given a "pass," and studios started to release "quad track" 35mm prints containing all three digital formats, plus a stereo optical analog track. Within a few years, most major studios releases were done this way, although many films continued to be released only with Dolby Digital, which eventually became the most popular format, both in filmmaker and exhibitor acceptance.

As noted earlier, digital cinema was initially a proverbial Switzerland of film sound, and this situation changed with Atmos, whose proprietary immersive format requires that a separate 5.1 or 7.1 PCM mix be included on DCPs. Auro-encoded printmasters, on the other hand, can play in any theater, and this summer for *The Amazing Spider-Man 2*, the 5.1 PCM mix was 11.1 Auro-3D encoded. When creating a 5.1 mix, the Auro Encoder adjusts the levels of the height and top layers, and these adjustments are "undone" when played back in Auro theaters.

However, the fact still remains that Auro-3D and Atmos, as the first two immersive sound formats in widespread use, are completely incompatible in philosophy, implementation and speaker layout. The situation is worse than had been the case with 35mm digital formats, and theater owners have the most to lose by investing in one system that is unable to play the competition's track.

In early 2013 the technology committees of the U.S. exhibition trade organization NATO (National Association of Theater Owners) and the European trade group UNIC (Union Internationale des Cinémas), along with DCI, joined forces in an effort to see to it that a "common immersive sound package" be utilized, as opposed to the 35mm quad-track solution of delivering all formats to all theaters.

Answering the call, SMPTE formed a special Working Group (TC-25CSS) to assist in this standardization effort. Auro Technologies and Dolby have both pledged to adapt to the agreed-upon open format. While Auro-3D is not currently object-based, their creative tools suite allows object-based mixes to be made, although it will not be in the same 5.1 or 7.1 PCM format as today. Also, the Barco cinema processors were designed with an upgrade path in mind, and 24 outputs, which presumably would allow the surrounds to be split into more zones.

Essentially, the goal will be for the metadata of any format's mix to be seen by any cinema processor's renderer, which is matched to the configuration file of a theater's specific speaker layout. Indeed, back at the mix stage, there have been mixes that were originally made in Auro or Atmos that have had panning data modified for the other format. The difference to the public would be how much the theater's system matches that of the mix stage.

One potential solution that has been presented is Multi-Dimensional

> Essentially, the goal will be for the metadata of any format's mix to be seen by any cinema processor's renderer, which is matched to the configuration file of a theater's specific speaker layout. Indeed, back at the mix stage, there have been mixes that were originally made in Auro or Atmos that have had panning data modified for the other format.

An Immersive Op-Ed From the Author

There's no way for me to write my opinions of the state of immersive sound in cinema without initially stating my prejudices.

First, I think that the 5.1 format that has been standardized since the coming of digital sound to 35mm release prints in the early 1990s is just fine.

In fact, I would almost take this opinion a step further and say that 4.1, with mono surrounds, is only a hair less better than 5.1. (Danger: One should stop operating power tools before reading the next sentence.) I believe that Apocalypse Now would be scarcely harmed with a mono surround mix. The greatness of that sound job lies in its ideas, which won't be diminished if a Loach doesn't precisely crisscross the theater or if the opening ghost helicopter were to start in all surrounds, as opposed to the right surrounds as it does today.

Second, the cost: All of the current formats ask theater owners to spend as much on this upgrade as they did with their original 5.1 systems—at least $50,000, most often closer to $100,000. For perspective, the cost of installing Dolby Digital processors over 20 years ago was less than $10,000.

Third, we must always remember that the drama of the movie is on the screen, within the proscenium. It's still a sheet on the wall, and even 3-D images do nothing to change this fact. If theatrical film sound must be extended beyond 7.1, there's plenty of gold to be mined in creative use of the remaining six channels available in a standard DCP, using them for specific speakers (especially left-center and right-center screen channels, or left-wide and right-wide just outside the screen) or additional surround arrays.

In this manner, all sound editing and mixing could be done without any elaborate object-based encoding and rendering. You would have a format that would "immerse" the audience to a degree as to be indistinguishable from Dolby Atmos or DTS MDA for 98% of the running time of most films.

I think that the missing 2 percent are not just an acceptable compromise, they're a desired one. What object-based mixing does uniquely—move sounds either down the length of the theater or to specific locations away from the screen—is precisely what I don't like. In the mid-'90s, some mixers took a few movies to get out of their system, putting silverware Foley or snare drums in the surrounds, and the thought of point-source sounds barking at me from the ceiling or walls is almost too much to bear. In fact, even the 7.1 format, dividing the surround tracks in two, bypasses my ken.

Now that I've said this, I admit that the toothpaste is out of the tube, and with hundreds of theaters worldwide putting in immersive sound systems, the question is not if immersive sound should be implemented, but how. First, let's look at the two approaches: Auro-3D, with its emphasis on height layers, vs. object-based Atmos and MDA.

In the multiple Auro-3D demonstrations that I have attended, I have not heard anything that I consider to be a radical improvement over 5.1. The most impressive parts of their test material were in sections recorded with their custom mic arrays, with various height levels. As good as these sounded, I think that custom mic arrays highlighting the strengths of Atmos and MDA would be much more impressive. Besides, as film sound history has shown us—such as with CinemasScope's three-track production recordings—literal reality is often not desired or practical.

The comparison between Dolby's Atmos and DTS's MDA is like Newton's third law of motion. To wit: Atmos' strengths lie in its rigid specifications that help ensure a match from mixing stage to commercial theater. What they have come up with sounds really good to my ears—bass-managed surrounds, the wide screen "proscenium" speakers, left-and right-center screen speakers in most theaters—this whole gumbo rocks and their unmatched industry support helps see that this is reality.

The downside? This quality is via by far the most expensive immersive system for theaters to install, frequently north of $150,000. And the content creators have to pay a license fee to Dolby, something that doesn't feel right in this era of digital distribution, where one can put a 5.1 or 7.1 mix on a DCP without any licensing fees or special equipment.

I've been friends with the folks from Dolby for decades, beginning first with Ioan Allen and Steve Katz in 1979, and following soon after that with David Gray and Doug Greenfield. My connection to film sound, first as a journalist and later behind a console, has evolved with these gentlemen and their colleagues at Dolby, and it's a relationship that I treasure. This being said, I no longer want to have to get Dolby's permission or gear to do a printmaster, and I'd prefer to put that money into additional sound editing or mixing time.

DTS on the other hand, is offering its MDA technology for free to filmmakers. Mixers will be able to start and finish their immersive mixes on their own, no different than the most humble 5.1 mix. Great, right?

Well, not so fast, Jackson. The "equal and opposite reaction" here is that, by having no "enforced" standards of how MDA theaters are set up, you can be assured that many will be installing two speakers overhead and calling it a day. And also calling it "just like Atmos." (I know this possibility pains my friends from DTS, and they would like to see SMPTE address speaker standards, too.) Of course, since Dolby has pledged to support whatever open format the industry is agreed upon, you could get lucky and have your MDA (if that is indeed the open format of the future) mix play in a theater originally equipped for Atmos. Or in a well-equipped MDA-only theater. But, "lucky" is the operative word, and if I'm at any level presenting a experience as "special" to the public, I want to be sure that they're hearing the film as I mixed it.

Whether or not a company is charging content creators (like Dolby) or not (like Auro-3D and DTS), they all see the real prize in convincing the home-theater-owning public of the need to keep installing extra speakers and purchasing new AV processors. While there are some unique potential benefits in object-based audio at home—I would love to watch an NFL game listening to the director and technical director calling out shots—I am not looking forward to reading members of the audiophile press comparing renderers.

Which way will I go in the future—stay 5.1 forever or "go immersive"? And if the latter, which format? I can't say right now, but I know it's going to be a real learning experience finding out.
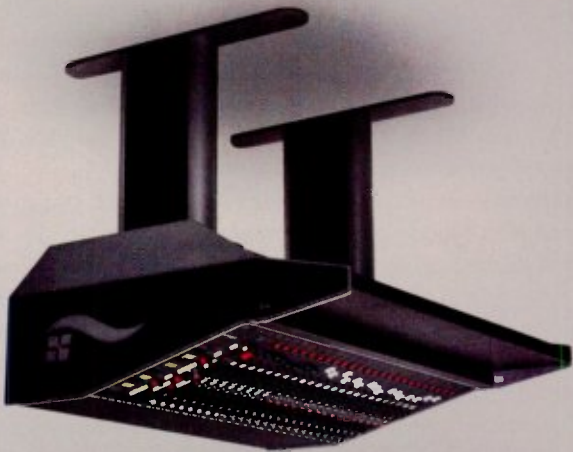
Audio by DTS. The company, which was originally known for its double-system digital theatrical format, split in two in 2009, with DTS keeping the licensing of consumer software and codecs. (The theatrical business was spun off to a new company, Datasat, which coincidentally manufactures the AP24 processors for Barco on an OEM basis.)

The intellectual property of MDA originally began at SRS Laboratories, before it was acquired by DTS. The "MDA Cinema Proponents Group," an informal alliance comprised of DTS and companies such as QSC, USL, Barco and Auro Technologies, has gathered to present MDA to TC-25CSS.

While MDA has not been used on any films, it has been tested in the industry, and version 1.0 of the code was released in early August, following up on specifications submitted to 25CSS months earlier. The SMPTE standardization process is famously long and drawn-out, and while the industry wants the format war to end, there's no reason that filmmakers or equipment manufacturers need to wait for any decision to be made to use MDA in the real world. (After all, Auro Technologies and Dolby didn't need to wait!)

Object-based like Atmos, MDA is being offered to the industry as an open format, with a SDK available to developers. As an open format, MDA would be license-free, and DTS would make available necessary software for digital audio workstations and console manufacturers. (Auro Technologies and Dolby have been providing similar support to filmmakers.)

Unlike Auro and Atmos, whose basic philosophies demand specific, scalable speaker locations and aiming (with Dolby going a step further in components and EQ), MDA is, by design, speaker agnostic. Indeed, there will be presumably much leeway in its implementation in theaters. For example, USL has come up with a cost-effective way for cinemas to upgrade by rendering the MDA mix to 13.1 channels of PCM files "offline," distributing those files to the media blocks of the servers in theaters. Rendering would take into account configuration files for individual cinemas.

Once an open object-based format is agreed upon, the next goal for theatrical presentation could be for the immersive file to downmix to 5.1 or 7.1 in theaters, avoiding the need to have a separate channel-based PCM mix on the DCPs. However, because this would mean that the downmixes are done without filmmaker intervention and control, it remains unclear if this is even a practical goal. Items like screen-to-surround panning would make downmix errors especially apparent.

FOLLOW THE MONEY ... TO THE HOME

The standard worldwide theatrical license for Dolby Digital in recent years has been around $11,000, giving filmmakers 16 hours of engineering support. A small increase this fall is anticipated, and the number of included engineering hours will increase at the same time. While the need for Dolby consultants on the stage increased greatly with the introduction of Atmos, as mixers become more familiar with the technology, their constant presence has been less needed.

The Dolby CP-850 cinema processor costs $33,750 [all prices in this articles are list] and includes the Dolby Commissioning Service: consulting on speaker layout and selection and doing the initial room equalization tuning. The unit can support up to 64 speaker channels, and at least one DAC3201 D-to-A, at $3,750 each, is needed for the first 32 outputs. The cost for each theater for retrofitting varies greatly, with some theaters north of $150,00 total. As of September 1, Dolby has over 560 Atmos screens worldwide, 175 in North America, and the total number of Atmos mixes at 120.

Auro-3D is free to content creators, and the Barco AP24 processor used in all Auro-3D theaters costs $25,000. As of now it has been installed in 215 screens, with the U.S. as the largest base, followed by India, and over 55 films have been released in Auro-3D. Up to this point, Barco is the exclusive licensee of Auro-3D for digital cinema, and Atmos is only available in Dolby's own units.

All of this is setting the stage for a conundrum waiting to happen: Everyone wants to promise a special experience for the public who leave their homes to go to theaters, yet all the formats are poised to expand to broadcast, Internet, home theaters, gaming, cars, mobile devices and pets. (The last is probably an exaggeration, but one never knows.)

One of the reasons that interoperable standardization is possible in theatrical films is that uncompressed PCM audio is used per the DCI specifications. Not only is it specified, there's much space on the hard drives that contain DCPs—the picture alone can take up hundreds of gigabytes.

Outside of theatrical presentation is a different thing, with media size (50GB on Blu-ray discs) and broadcast bandwidth limitations. Hand-in-hand with this is the decision as to which lossless (such as Dolby TrueHD or DTS-HD Master Audio) or lossy (Dolby Digital Plus or DTS-HD) codec to use.

Auro-3D, in formats such as Blu-ray would be able to use its original 5.1 theatrical PCM printmaster, where Atmos and MDA mixes will presumably need to have separate home video immersive printmasters created to fit in the smaller bandwidths.

Dolby and Auro Technologies have taken their first steps to get their immersive tracks in AV processors. Auro Technologies announced their own Auro-3D Auriga unit at this year's CES show, and they have signed up McIntosh, Datasat and Lyngdorf, among other companies, to bring Auro-3D to the home.

Dolby's serious push for Atmos at home began in August, with demonstrations around the U.S. to the consumer audio press. Atmos for home theaters is scaled down (from a maximum 64 speakers in theaters) to 24 floor speakers and 10 overheads. They expect that most homes will have no more than four overhead speakers, and Dolby has anticipated practical mounting issues by designing "Atmos enabled" speakers that fire up at the ceiling.

DTS will clearly be making a big push for MDA's use in all media; donating it as an open, free format for cinema exhibition usage has to pay off some time. They and Dolby have dominated the licensing market for home audio for decades now.

Object-based audio at home will allow manipulations like turning announcers off, and just listening to the director calling shots in the truck. Or the pit crews at NASCAR. Or just the immersive stereo sound on the field. The possibilities are endless.

With the latest cinema sound format—now, immersive audio—leaving multiplexes and going to the home, the "natural" order has been restored. The prior "standard" format, 7.1, began in home theaters before heading to cinemas.

At the end of the day, everyone wants to make more money, and it will be up to the collective votes of filmmakers, exhibitors, theatrical moviegoers and consumers as to which will prevail. The extreme cost and paradigm shift involved on all fronts makes this perhaps the most difficult transition in film history.

DENNIS SANDS' SOUND WAVES SB

A PRIVATE DOLBY ATMOS MIX RETREAT

BY BARRY RUDOLPH

Four-time Oscar-nominated music mixer Dennis S. Sands has taken his extensive experience and expertise in orchestral recording and film score mixing and brought it all to bear in the creation of his magnificent home-based facility, Sound Waves SB, the first private mix facility we know of to be equipped with the Dolby Atmos system.

With an engineering/mixing resume that includes landmark films such as *The Shawshank Redemption*, *Good Will Hunting*, *Independence Day*, *The Horse Whisperer*, and *Who Framed Roger Rabbit*, plus current blockbusters such as *Captain America—The First Avenger* and *Godzilla 2014*. His Oscar nominations for Best Achievement In Sound Mixing include *Forrest Gump*, *Contact*, *Cast Away* and *The Polar Express*. He has six additional C.A.S, Golden Reel, and Satellite Award nominations, plus a Grammy for Best Score Soundtrack Album for the film *American Beauty*.

Always looking to bring his art to a higher level, Sands was recently introduced to the Dolby Atmos immersive platform at an L.A. dub stage, and immediately recognized its potential. He went home and re-outfitted his al-ready impressive studio, enabling him to deliver a film score in Atmos, 5.1, 7.1, and stereo.

The facility itself has an impressive pedigree, too. Originally built by action-film director Andrew Davis (*The Fugitive*), the studio, now known as Sound Waves SB, was designed by Tomlinson Holman and outfitted from top to bottom by RSPE Audio Solutions of Los Angeles, in conjunction with Ron Lagerlof of Visioneering Design. The SB stands for Santa Barbara, Sands' beautiful and peaceful getaway from the rush of Hollywood

Here's what Sands has to say about his new mixing home.

What kind of experience can clients expect when they come to Sound Waves SB?

Everyone who comes here is blown away by it. First, it's extremely com-

Sands' monitoring system is anchored by three Meyer Sound Acheron 80 screen channel loudspeakers and two X-800C high-power and X-400C cinema subwoofers each. Perimeter and overhead reinforcement comprises 14 HMS-10 and 10 HMS-12 surround loudspeakers, while a Galileo loudspeaker management system with one Galileo 408 processor supplies drive and equalization. The system was designed and installed by Ron Lagerlof in consultation with Andy Potvin of Dolby Laboratories.

fortable and very private. It's relaxed, it's beautiful, and the facility is unique in the music world—there's nothing like it anywhere. It's also extremely well equipped—with great sound and a product that translates very well.

Plus, we have great projection: a Christie Digital Systems projector, a 12" by 20" screen, and a full Meyer Sound cinema sound system. It's a very large space (42x36 feet, with 16-foot ceiling), and both RSPE and Visioneering were initially contracted by Andy (Davis) and have been involved with this facility since its start. RSPE supplied the equipment, Ron Lagerlof the technical expertise to interface it all—and they both provided the same capabilities for me.

Primacoustic... better design, better

"The ease of install really allowed us to experiment with placement and with the quality of the treatments, we achieved the sonic balance we were looking for!"
~ **Tommy Lee**
Founding member - Mötley Crüe.

"Being able to fine-tune a room on site makes all the difference. The Impaler mounting system make the panels easy to install and let you make adjustments without trashing the surface. It works!"
~ **David Rideau**
Engineer/producer - Janet Jackson, Sting, TLC, George Duke and Jennifer Lopez.
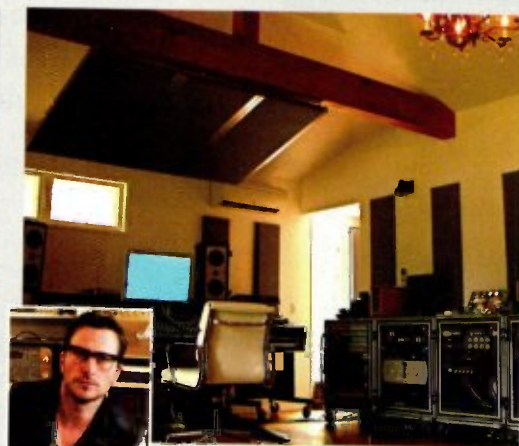
"The Primacoustic is up and kicking butt at my new studio in Santa Monica. I love the way the control and tracking rooms sound now... and so does everyone that records here!"
~ **Butch Walker**
Engineer/Producer - Avril Lavigne, Fall Out Boy, Pink, Sevendust, Hot Hot Heat, Simple Plan, The Donnas.

"I love the way the control and tracking rooms sound now...
and so does everyone that records here!" ~ Butch Walker

And finally, we're in Santa Barbara, which is a resort destination. We have a huge lawn here with benches, so we can sit outside and have lunch. There's also great food and luxury hotels—Santa Barbara is all about accommodating visitors.

Let's talk about Dolby Atmos. Does it change the way you mike an orchestra?

I'm experimenting a little now, but I haven't found anything that I would do radically different...as yet. Just to backtrack a bit as to why I put Atmos in my facility: at the end of December 2012, I was working on a project called Oz the Great and Powerful. The director contacted me and said that they were going to release the movie in an immersive format and wanted me to listen to it, because they wanted me to do something for the score I said okay, but I was very cynical about it. I thought, "Boy, 5.1 or 7.1 is pretty nice, why do we need another format?"

So I went to a dub stage in L.A. where they had an Atmos demo set up. The first time I heard this stuff—and no one was more cynical than I was—the first time I heard it I was completely blown away. "Immersive" is a very good term. The sound isn't thrown at you; you're in it, in a great, great way. Not only that, but the clarity, the depth, the accuracy of the movement of sound, and the placement of sound in the 3-dimensional environment, was stunning—something I'd never heard before, really.

So I started to look into it. But at the time, outside of going to a dub stage, I had no way of hearing it, and certainly no way of mixing music

with it. There was just nowhere to go, and now I have the only dedicated music-mixing facility with Atmos capability. There simply are no other facilities.

What about the overhead speakers? How do they figure into the way you mix?

I look at the overhead speakers the way I look at a subwoofer. I think what makes the overheads work is when you use then occasionally. If you use them all the time, they'll disappear. Subwoofers are the same thing. If you have a low-frequency thing going on all the time, it loses its effectiveness. When the overheads are there occasionally, then it opens up—it's impressive and beautiful.

One of the great things about Atmos is that the surround channels are full range and of equivalent volume to the screen channels. All speakers are set to the same standard level, plus, the surround channels are bass-managed, which means I have subwoofers that enhance the low frequencies on the surround channels and the ceiling channels. You now have full audio range throughout the entire theater, which you never had before. It's a totally different soundscape. And because it's so accurate in terms of how you can position various musical elements in a space, it's really an incredible experience for the audience—a new experience.

So Atmos leaves a lot more room for Foley...

As well as the dialog. Instead of being locked to the screen, you can

spread the width of the music to the walls and still have full frequency-range capability. It allows more space for dialog, more space for sound effects, and you can still keep the music happening—there's just less competition for space. In essence, you can move sound off of the screen.

Do you think Atmos will change the way composers work?
I hope so. What I would love to see happen is for composers to a) get excited about this format, and b) think about it when they're composing, so they actually start writing for this kind of experience. People who are creative, I think, will find this a wonderfully additive tool.

What else do you have here at Sound Waves SB?
One thing that is substantially helpful to me, especially in the interface

of my Atmos mixing, is the Euphonix System 5 MC that I got from Russ at RSPE. It's a control surface that's just absolutely spectacular and is an integral part of Atmos control for our studio.

When I moved here, I had a Euphonix CS 3000, a 96-input desk, which is digitally controlled analog. I looked at several digital consoles, but they are limited to a 96k maximum format. I do all of my orchestral and/or acoustic recording at 192k. The difference from 96k to 192k is stunning. The quality, the dimensionality, the depth of the sound, the content, the richness, is stunningly better at 192k. But the limitation of digital consoles is that everything you do, whether you set up a channel, effects send, EQ, compressor, whatever, draws on the available DSP. From a practical standpoint, you cannot go above 96k on a digital console, so I chose to keep my analog console to allow for the highest possible quality.

Because I kept my CS 3000, I can remain in the 192k format, which means the quality is substantially better, but I need to access my other systems. I have four systems online when I mix: one for prelay, one for orchestra, one is the mix rig, and the fourth is a dedicated video system.

With the System 5 MC, everything is right in front of me. At the push of a button I access whatever system I want, do my moves, save 'em, push a button, go to a different system, it's all right there—just fantastic.

Editor's note: While he may be at ease with miking horns, Sands is not the kind of person who's comfortable with blowing his own. So we asked Gary Lux, former head scoring mixer at Universal Studios, to close the interview.

Lux: I recently visited Dennis at his studio in Santa Barbara where he demonstrated a Dolby Atmos mix he had done for the then-upcoming movie Godzilla. I've mixed a tremendous amount of music in 5.1, but hearing his Dolby Atmos mix blew me away, as it opened my mind to the possibilities I had tried to achieve in 5.1 for so long. But, the real magic to Dennis, besides the talent and his wonderful ear, is how incredibly nice a man he is, which makes him one of the greatest music mixers this town has ever known.

For more information on Dennis Sands and Sound Waves SB, visit www.soundwavessb.com. For more information on RSPE Audio, visit www.rspeaudio.com.

CAROLE KING IS *BEAUTIFUL* ON BROADWAY

BY ERIC RUDOLPH

Jessie Mueller won a Tony for her portrayal of Carole King

Photo: Joan Marcus

Carole King burst onto the music world from no-where in 1971 with her huge-selling album, *Tapestry.*

Well, yes and no.

Years before The Beatles disrupted the business, a teenaged Carole (Klein) King was a staff songwriter, penning pop and R&B hits in an office cubicle.

With songwriting-partner and husband Gerry Goffin, King wrote some of the biggest hits of the era—on salary and in a space big enough for an upright piano, bench, chair, two people and little else, at 1650 Broadway ("the famed rabbit warren where white people wrote songs for black singers," quips *Variety*).

The unlikely path and duality of Carole King's career—from teenage-Brooklyn-mom and Times Square-hitmaker to divorced-Laurel Canyon-zeitgeist-defining-megastar has made great copy since she swept the Grammys with the inescapable *Tapestry*.

Now her tale is *Beautiful: The Carole King Musical*, a Tony-winning Broadway smash-hit spanning 10-plus years of the artist's life.

It starts in the late '50s when a precocious teenager begs her mother to let her travel from Brooklyn to Times Square to pitch a song, and it concludes with the multi-platinum-selling artist's triumphant gig at Carnegie Hall.

Beautiful also tracks the initially similar careers of songwriting team Cynthia Weill and Barry Mann ("You've Lost that Lovin' Feelin'", "Walking in the Rain," "We Gotta Get Outta This Place"), who were Goffin/King's cubicle colleagues and friendly competitors.

Beautiful won two 2014 Tony Awards; star Jessie Mueller [Carole King] won for best actress in a musical and sound designer Brian Ronan won for best sound design for a musical. The show was nominated for seven Tonys.

Sound design is indeed a big part of what makes *Beautiful* work, dramatizing the contrast between what the highly personal songs mean to their writers and how they're transformed under the white hot light of show business.

Beautiful also uses the increasingly sophisticated changes in audio production styles—from the simple, tight 45 rpm-single sound of the late '50s/early '60s to the processed, expansive multitrack recording and richer live sound of the early '70s—to help tell the story. As Carole King travels from Brooklyn to Times Square to a world stage, the increasingly richer audio quality tracks her journey. "The sound design follows Carole King dramaturgically, it is part of the storytelling, approximating the sounds that were available to her," Ronan says.

When the scared-but-bold teenaged King auditions for music publisher Don Kirshner (who'd had a hit with her high school friend Neil Sedaka), sound designer Ronan begins to sonically emulate the time and place of the story.

"We begin with Carole playing an upright piano in Kirshner's office," Ronan explains. "It's actually our conductor Jason Howland playing, as he watches Jessie Mueller via a hidden video camera, following her hands. Sonically, the effect is true to the script—an acoustic piano in an office."

It sounds real because it is...sort of. The piano sound is indeed emanating from the onstage piano, from speakers hidden inside the prop instrument: Howland's piano playing gets there via a wireless in-ear monitor rig. Any time an onstage piano, amplified guitar, TV, radio, etc., is playing, the sound is coming mainly from the onstage prop source via this technique, to focus the listener to the stage and enhance the verisimilitude.

Significant Broadway-style dramatic license is then taken as the rest of the band joins in, to support King's description of how she hears her song sounding on a record.

However, while the music swells, the amplification dials down. "We're leaving sections of our band muted and allowing the acoustic instruments—mainly reeds and brass—to flow naturally from the pit to give an un-amplified, organic feel," says associate sound designer Cody Spencer. "We're keeping it simple, like she's playing by herself, and the band is [coming from her imagination]. We're showing how she thinks her music will sound, and also transitioning to bigger production numbers in the show."

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That scene is book-ended with a recorded track representing King's first song sale and hit record, "It Might As Well Rain Until September." This time the sound goes from big to small, evoking the way teen pop music sounded in the pre-Beatles era.

"An actor plays singer Bobby Vee while the pre-recorded music suggests a commercial pop song in 1962. We transition from the pre-record coming out of the whole P.A. and crossfade it down," dramatically, from all those big speakers, "to a transistor radio carried by an actress playing a student," says Ronan.

The effect is totally appropriate to the time; like hearing music through tiny cellphone speakers, tinny-sounding transistor radios were how teenagers heard Top 40 radio in mobile mode then. It is also quite convincing, as the sound actually comes from a tiny computer speaker in the prop radio the actress holds as she sashays across the entire stage, organically phasing the sound.

The transistor radio isn't the only retro tech in *Beautiful*. Vintage mics abound onstage but are mostly props; the actors almost always use their hidden head mics. Actual working vintage mics were tried in a few scenes out of town, "but they sounded really, really bad," notes Ronan wryly.

One powerful scene that called for a vintage mic approach instead got a completely anachronistic treatment due to audio and theatrical demands.

Near the end of Act I, when the staff-songwriting machinery nears full speed, Barry Mann (Jarrod Spector) gives the classic Mann/Weill hit "You've Lost that Lovin' Feelin'" a quiet songwriter's reading, intimate and full of youthful angst. Then actors portraying The Righteous Brothers come dramatically into the spots, singing the classic hit with the full band, approximating record producer Phil Spector's legendary Wall of Sound, with major volume, reverb and compression. Their starkly contrasting performance is slick, sophisticated soul music.

The Righteous Brothers characters' hip attitudes, slick hair, shiny skinny suits and vocals are all spot on, but audio purists may howl; the two singers are using modern Sennheiser wireless handheld mics. It is a big compromise for the sake of a dramatic, punchy vocal sound, contrasting with the preceding quieter solo vocal of the same song, and it was a reluctant compromise for Ronan.

"Out of town we started with prop vintage mics, but really using the singers' head mics," he says. "But you just can't get the warmth and proximity effect that's so familiar to everyone that way. The Sennheisers are, of course, not true to time, but we also needed no wires, to get on and off stage cleanly."

FRONT OF HOUSE

The *Beautiful* system runs 132 inputs, with 40 channels dedicated to head mics, mostly Sennheiser MKE-1s. The band gets 64 channels. The DiGiCo SD7T control surface, Spencer says, "really helps us; the tube emulator in each channel helps warm up and cut through, and the Dynamic Compression on every channel helps with some of the actors us-

2014 Tony-winning sound designer Brian Ronan at The Stephen Sondheim Theater, FOH, with the multi-level stage set that represents the 1650 Broadway offices where Carole King, Gerry Goffin, Cynthia Weil and Barry Mann wrote songs for publisher Don Kirshner, in the background.

Photo: Eric Rudolph

ing side-rig head mics, instead of the preferred lav placement at the hairline. Side rigs makes it hard to get the chesty sound, so we're easily able to warm them up."

The Stephen Sondheim Theater, one of the newest on Broadway, also helps as it has only two levels and is fairly wide. "It is a great-sounding house," says Spencer. "Being new, it was built with some soundproofing and with acoustic paneling up top, and used a lot of porous material, not hard plaster."

The main speakers are a center-hung line array of 12 L-Acoustics dV-DOSC and two Meyer Sound CQ-1s. The big CQ-1s "are aimed straight to front orchestra seats with just band, no vocals, to fill the front rows that don't feel the band that much in those 'money seats,'" Spencer notes.

Two L-Acoustics dV-SUBs are right and left just outside of the orchestra seating area and two more are up near the spot booth. More than 50 Meyer speakers do point source duty as well as foldback, with MM-4s and UPJuniors hidden onstage and in lighting towers as monitors. Foldback mix is band only, no vocals, with the actors feeling the house for their vocals. There are no in-ear monitors.

Meanwhile, even after The Beatles sideline salaried songwriters, the hits keep on coming for Goffin/King and Mann/Weill.

Later in Act II there is another dramatic audio transition where we initially hear a full-blown version of Goffin/King's "Pleasant Valley Sunday," a major hit for the original Prefab Four, The Monkees, mocking suburban life.

The song is placed to dramatically put Goffin and King in their new suburban New Jersey home—a very big move for these lifelong city kids—as they audition this song for Kirshner. "The beauty of theater gives me license to start in full-tilt '60s Hi-Fi sound," with a wide left-right pan for the band and vocals, and kick and snare right up the middle, for a big, full sound true to the time. "It's an unnatural, commercially-viable and chronologically-correct sonic image," adds Ronan. This sound also reflects the new suburban affluence of King and Goffin.

As the lights come up the big, full production sound of the song crossfades to Goffin and King performing "Pleasant Valley Sunday" as songwriters in their suburban living room. "We begin to pan back the separation to a more natural centered sound and peel off the mics on the drums, until we're just on piano," Ronan explains.

"It's one of my favorite moments because it marries the time, scene and sound in a way that allows the audience to take this mini journey," from big show-business gloss to the simple home life of the writers, "unaware that the sound design has been manipulating them."

Story-wise, all seems idyllic, with a second child for King and Goffin. Their creativity (and that of friends Barry Mann and Cynthia Weill) is flowing freely, and as writers they're beginning to catch up with the changing times.

The 1960s sitcom-like interior of the suburban home is ironically the

setting for a gritty Mann/Weill song, "We Gotta Get Out of this Place," with the Mann character singing and strumming an electric guitar, all while seated with his striking blonde wife on a comfy sofa in the 'burbs. As noted, the guitar sound comes mainly from the stage via a pit guitarist, thanks to a specially modified onstage Fender guitar amp.

All seems well until Kirshner explains that he loves "Pleasant Valley Sunday" and it will be perfect for his song-hungry new project: a weekly half-hour TV show that's just like *A Hard Day's Night*...just without the Beatles.

Goffin, who has long chafed at writing for teenagers and feels the pressure of The Beatles and Bob Dylan, explodes that he's been reduced to writing songs for a sitcom. That foreshadows the final dissolution of the shaky marriage and King's transformation into a solo megastar.

From here the sound grows fuller and richer, with more sophisticated use of reverb, compression and modern mixing techniques. The songs made so famous on *Tapestry* take over, and King leaves the Jersey sub-

L-R: Jeb Brown as Don Kirshner, Jake Epstein as Gerry Goffin, Jessie Mueller as Carole King, Jarrod Spector as Barry Mann, and Anika Larsen as Cynthia Weil in *Beautiful - The Carole King Musical*.

urbs for Laurel Canyon in Los Angeles to write and record solo, after her marriage ends. She makes *Tapestry* and LP sales history, defines an era and inspires generations of female singer/songwriters.

As for the real Carole King, she stayed away from the Broadway version of her life for months.

"Ms. King wouldn't come near the show. Couldn't get her to do any press or promote it in any way. She's kinda shy," says Ronan. "She finally bought a ticket, disguised herself and watched the show."

For the curtain call, Jessie Mueller sings "I Feel the Earth Move." "Jessie uses a wireless handheld mic, so we can get good and loud over the clapping and singing along," Ronan recalls of the night. "Carole King was brought onstage with our spare handheld mic and she just jammed with Jessie, no rehearsal, no soundcheck."

The star did reach out to the sound designer, however. "She left me a lovely message on my phone when I got nominated for the Tony. I did get to meet her briefly on Tony night," Ronan notes, concluding his *Beautiful* show business story. ∎
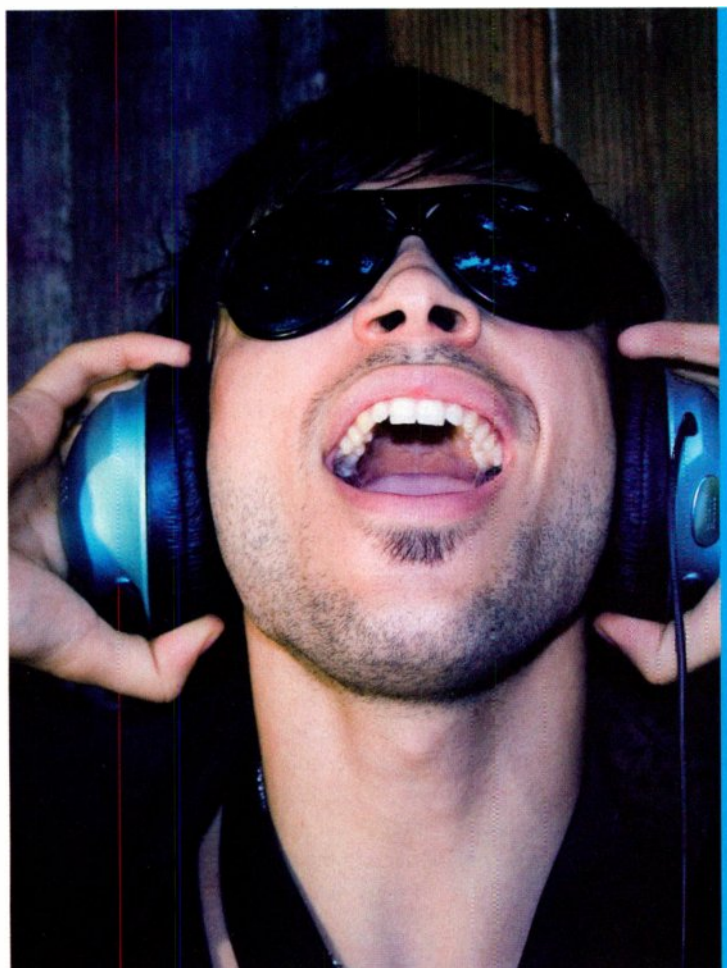
CRAS Rolls Out Broadcast Audio Curriculum

BY TOM KENNY

Robert Brock, Fred Aldous and Kirt Hamm at the Studer Vista 9 in the CRAS Broadcast control room.
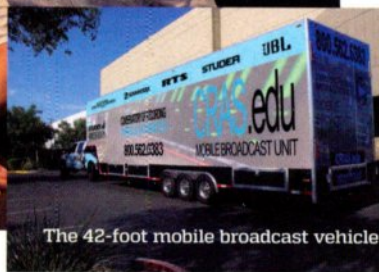
The 42-foot mobile broadcast vehicle

Over the past 10 years, with the decline in the number and staffs of commercial studios nationwide, recording schools have added to their curricula and students have found new opportunities in audio—videogame sound, house of worship sound, corporate sound, film sound, etc. But there has been very little attention to broadcast sound, by any of the leading schools, which seems a bit odd when you look at the size of the broadcast landscape.

In early 2014, the Conservatory of Recording Arts & Sciences in Tempe, Ariz., one of the nation's leading audio schools, launched its Broadcast Sound component. "My goal for all students is employability," says CRAS administrator Kirt Hamm. "Most come for music, but we want to expose them to other areas of the audio industry, too. Over the years we have spent more time in post-production for video, game audio, live sound and concert sound, the business side of music. The broadcast side came about after Fred Aldous approached us with what he did and asked if we had ever thought of a broadcast curriculum. He was pretty persistent." [laughs]

Aldous, audio consultant and senior mixer, NFL and NASCAR, at Fox Sports, lives in Gilbert, Ariz., and is one of the top live sports mixers in the country. He had come up in the industry under the mentorship of the legendary, late sports mixer Bob Seiderman. And he had grown at Fox under the direction of Chairman David Hill. In 2005, he taught a guest class at CRAS, then a few more the following years. He took a nighttime course in Pro Tools and Logic Pro, met Robert Brock (who was then digital recording department director and now is director of broadcast audio) and proposed the development of a dedicated curriculum. There was a need in the

industry to develop the next generation of audio mixers, he told them, and nobody was filling it.

"There has become a void of qualified mixers in broadcast, sports mixers especially," he says. "The shows have become so complex, and we're not onsite as long for each event, so there is no time to teach anybody on the job. We're in and out and there's no time to mentor. At the same time, the networks all want people with experience, but how do you get that experience?"

Aldous invited Hamm and Brock to NYC for a Sports Video Group conference.

"We were pretty surprised about the conversation the audio side was having at SVG," Hamm recalls. "They were clamoring for this need to be filled, which energized Brock and I. Then we went to the Giants NFL game and we saw the setup, the complexity. It was a great experience and exposed us to the fact that at the one football game they had 12-plus guys for audio-only." They returned to Tempe and started developing a curriculum, writing their own textbook. At the same time they researched equipment and workflow to outfit a new studio and 42-foot mobile truck.

The three-week intensive Broadcast Audio component comes in Cycle 10 of a 12-Cycle program, so students come in well-grounded in signal flow, audio basics and recording/mixing. On the first day, a countdown timer is set for three weeks out, when the cycle culminates in a live-to-tape show run entirely by students. The goal is to expose students to broadcast audio, not make them A1s—there isn't enough time for that.

"There were two main areas we identified that would be unique to broadcast audio," says Brock. "First is a true sense of pressure, that this show is not going to be delayed even 5 seconds because they're not ready. It's all about your preparation. Then we put them in an environment where they are have to respond to things like director calls—'Roll A, track full'—what does that even mean? Our students get that exposure to the language, they learn how to selectively listen to a director while doing three other things. The other aspect is communication systems. Comms in a recording studio is pretty simplistic; comms in a broadcast scenario is absolutely crazy."

"Comms is pretty much the heart and soul of a broadcast," adds Aldous, who might be responsible for up to 500 lines during a Super Bowl. "If the director can't talk to cameras, or producer can't talk to tape operators or the audio mixer can't listen to direction—if the comm systems are down, there is no show. I think we are one of the few if not the only schools that is teaching basic communications, as far as how to set up an IFB, a PL, point-to-point. That is a foundation of what we are doing.

"We're also psychologists," he continues, "and we work with announce talent much like a producer works with an artist. So we teach that. Also, how to interact with other departments in the production truck. You have to have rapport with the tape room, the graphics department, the technical director and your own audio crew. You have to be harmonious with your broadcast team. In live television, if you're not ready at 1 o'clock when that ball is kicked, you're being left behind."

To launch the program, CRAS built nearly identical broadcast studio and mobile rooms, the only real difference being that the 42-foot gooseneck trailer has a 15-foot wide expando, and a Studer Vista 5 console. Studio E on the Gilbert campus has a Vista 9 with identical I/O, just a slightly different surface. Both have RTS comm systems, Studio Technologies Model 210 announce consoles and IFB, and 5.1 JBL LSR 6325 monitoring.

The decision to go with the Studers came after research into every major manufacturer of digital consoles. "We were looking for a real relationship with a manufacturer," Hamm explains, "not just someone to sell us a console. We wanted a recognized platform, and we wanted something we could teach on. We were introduced to Michael Franklin and Katie Templeman-Holmes at Studer, and they got it. They saw benefit in our expertise of curriculum development and design. That's how we started the Studer Broadcast Academy."

Brock helped design a professional curriculum for the Vista Series in conjunction with Studer, and to date more than 1,000 users around the world have been certified through the online program.

Besides their class requirements, students have also been out with the mobile vehicle to two music festivals, the Potluck Conference, the Center for the Arts, and there are more excursions planned. The first class of students are now graduating, so the challenge is to place them in (required) internships, establishing all-new contacts with a whole new industry. No doubt there will soon be CRAS graduates helping bring Saturday afternoon college football over a station near you. ∎

other markets? When we looked at this, we knew we had *Spiderman* coming down the pipe. We had been talking to both Barco Auro and Dolby Atmos about their formats, and we felt we were ready to make that leap. Other studios had invested in it, and we wanted to as well.

"So we decided at that time to put Atmos and Auro in the same room," he continues. "Some of our talent wants to mix native Atmos, others Auro. Some want to hit the most theaters in 5.1, attacking the object tracks later. The thought was, let's put it all in one theater so we have the ability to mix 5.1, 7.1 and enhance to an Atmos or Auro format. Or the reverse. Either way, we're staying in the same environment, the same room, and we don't have to shift and we don't have scheduling conflicts. From that standpoint, it made the most sense to put them both in the Holden."

McCarthy has had quite a colorful career in sound. An industry legacy, with a father as a picture editor (and later EVP Worldwide Post-Production, Columbia Pictures) and uncles in cinematography and sound, he started in 1975, at age 21, in a sound effects library, followed by a brief stint in picture editing, then Foley walking, then Foley editing and dialog editing, falling for sound as a storytelling device. He left MGM to work on Heaven's Gate, and went independent as a sound supervisor in 1981, opening The Sound Choice in Burbank.

Michael Kohut, then a mixer and in charge of motion picture sound at Sony, approached him a decade later about running motion picture sound editorial. McCarthy accepted, continuing to supervise while running the department. In 1993 he won an Oscar for Francis Ford Coppola's *Dracula*. A year after that he accepted a vice president job and gave up his creative day to day. It was hard to do, he says.

His companion on this month's cover, Bill Baggelaar, senior vice president for Colorworks and Sony Post Production, joined the company in 2011 after years at Warner Bros. in both production and post-production technology. With a degree in computer science, he had been involved primarily on the picture side, logging years in animation systems, then feature post-production, followed by years of digital intermediate, among various other responsibilities. He came to Sony to help push 4k integration and advanced workflows, and now also manages the sound engineering department.

"My first experience with immersive sound began about five or six years ago, with demonstrations, both within the industry and our own company," Baggelaar recalls. "It was certainly ear-opening. It was a revolution in sound that felt different, that you could sense had great potential.

"So when we started talking about putting immersive sound into the studios here, deciding eventually on the Holden, we talked about how we could satisfy all the needs we have, given the scope of the studio, our distribution, and looking at things more broadly," he continues. "It was really a rethink about what we wanted to do in the environment, rather than just put up some speakers. We tried to make sure that we engineered it in such a way to give the mixers the best tools, ones that they could quickly learn and get into.

"At the same time, we also put in 4k projection, and we put in new screens, so you could have the fully immersive experience. We wanted to be able to provide any configuration that might be needed creatively, and we wanted to focus on the Dolby and Auro formats first and fold down to the 5.1 mix. Thinking about object-based audio from the get-go was a new way of doing it, and I think everyone rose to the challenge."

While McCarthy and Baggelaar sit on the *Mix* cover, they are both adamant in pointing out that it was a facility-wide effort to update the Holden—from Brian Vessa in digital audio mastering, and his exhaustive research into the competing formats and decade-long work with the Digital Cinema Initiative and various SMPTE working groups, to Bill Banyai and the sound engineering team with their tireless work in the three-month installation, along with countless others. They worked with Harrison on the updates to the tools and software of the 320-channel MPC3D console, and with JBL on additions to the custom cinema monitoring system.

"There are different pannings and plug-ins between the formats, and we have some crossover in the speaker configurations, but the mixes can all be accomplished in the same room," Baggelaar says. "The desire was to offer filmmakers and sound creatives the maximum amount of flexibility."

"Today, with rapid advances in technology and the way we function, if we're going to be successful and sustain our businesses, staying within budgets, we have to be flexible in the way we operate," concludes McCarthy. "There have to be multiple types of workflows that individual filmmakers can embrace and be comfortable with. Today, these immersive sound formats are beyond belief. They're that good. We can't determine what will happen in the future, but based on the advancements of the last 20 years and how rapidly they are changing, I can't imagine what we will have 20 years from now." ∎

MDA

FROM DTS

FREE, FLEXIBLE, SCALABLE
OBJECT-BASED AUDIO

World Radio History

FIELD RECORDERS

AUDIO OPTIONS FOR FILM AND VIDEO PRODUCTION

BY MATT GALLAGHER

Location recorders have long been a staple of the film industry, all the way back to Stefan Kudelski's introduction of the Nagra portable tape recorder in 1951—providing a compact, straightforward solution for capturing high-quality audio in the moment with the ability to interface with other essential audio and video devices. Mix surveys the world of portable recorders designed for capturing dialog and sound effects out in the field, or on-set, for professional-level film and video projects. Here, we present a sampling of these products, including 2-channel and multichannel units, ranging from proven, trusted and familiar models to some newer offerings.

AATON CANTAR-X3

Aaton's Cantar-X3 follows in the footsteps of its predecessor, the Cantar-X2, having been designed by the same team. Cantar-X3 inherits the ergonomics of the X2 and its microphone pre-amplifiers, which have been redesigned to improve filtering possibilities and reduce the noise level. It also offers Audinate Dante IP networking as an option, and a display-based menu control that was designed by Transvideo, a designer and manufacturer of LCD flat-panel monitors

in France. The Cantar-X3 is further optimized for location work due its avionics aluminium machined housing and its military grade water- and dust-proof durable coating. The Cantar-X3 features extended linear faders, smooth rotary knobs and silent switches.

FOSTEX FR-2LE

Manufactured from high-grade materials and specifically designed for professional location recording, the Fostex FR-2LE records 24-bit/96kHz BWF files, as well as MP3 files, to a Type II CompactFlash card. On an 8GB CF card, it can record 12 hours at 16-bit/44.1kHz. The FR-2LE is equipped with two phantom-powered XLR microphone inputs (XLR-phone combo) complete with precise control over the recording level, as well as high-quality microphone preamps and trim control. Analog out is offered on RCA connectors. It features a 1 take = 1 file recording system to eliminate overwrites, and it has a 2-second pre-record buffer.

NAGRA SEVEN

The Seven is a 2-channel, 24-bit digital recorder fitted with an internal 16GB micro SD flash memory, equipped with AES-42 facilities for digital microphones and offering sampling fre-

quencies up to 192 kHz. It offers a selection of internal optional circuit cards that allow it to be adapted to 3G and Wi-Fi, SMPTE/EBU timecode or ISDN communication. In addition, there are software options for internal audio editing and audio compression (MPEG). The Seven offers two analog audio inputs and a single stereo digital input. It will record in PCM linear audio at sampling frequencies up to 192 kHz or in MPEG-1 layer II compressed audio formats if the compression option is installed. The inputs are equipped with traditional Nagra mic preamps for dynamic and phantom-powered microphones, and offer switchable sensitivity settings. The analog line input will accept up to +24 dB.

ROLAND R-88

The Roland R-88 includes various timecode modes and slate tone/mic for memos. The stereo mix output on the R-88 is designed for lo-

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cation sound recording applications enabling a stereo mix back to the camera to record reference audio, or providing a stereo monitor output while simultaneously recording individual tracks. With eight discrete outputs users can also embed the audio tracks into a SDI video feed by using a multichannel SDI audio embedding device. The R-88 also offers AES/EBU digital I/O to digitally record from or send to a mixing console or other digital source. The R-88's built-in mixer is equipped with a 3-band EQ enabling a stereo mix to be monitored or recorded simultaneously along with each discrete audio track. The R-88 has a built-in 10-input/8-output USB audio interface for multichannel recording.

SONY PCM-D100

The PCM-D100 is a high-resolution recorder with two-position (X/Y or Wide) electret condenser microphones, 32 GB of internal flash memory and a SD-XC Card slot. Recording formats include linear PCM (at 192, 176.4, 96, 88.2, 48 and 44.1 kHz); DSD (2.8224 MHz) and MP3 (320 and 128 kbps). Additional playback support is provided for FLAC, WMA and AAC files. The PCM-D100 offers comprehensive signal processing features for location recording including a limiter and low-cut filter. The PCM-D100 also includes a 5-second pre-record buffer and cross-memory recording function. The PCM-D100 is constructed of lightweight aluminum and uses conventional AA alkaline batteries. The recorder includes a USB high-speed port, digital pitch control, dual path digital limiter, Super Bit Mapping and A-B repeating capability.

SOUND DEVICES 970

Sound Devices' half-rack, 2U model 970 records 64 channels of monophonic or polyphonic 24-bit/48kHz WAV files. Connections include 64 channels of Ethernet-based Dante, 64 channels of optical or coaxial MADI, eight channels of line-level analog and eight channels of AES digital. Any input can be assigned to any track, and it supports 32-track recording at 96 kHz. The 970 records to any of four attached drives: two front panel drive bays and two rear panel e-SATA connected drives. With four available drives, material can be re-

corded to multiple drives simultaneously for backups. Drives can also be set to record sequentially, allowing continuous long form, high-track count recordings. The 970's built-in Ambient timecode generator allows it to operate as either the master clock or be jammed to external devices cameras and other sources. All common production timecode rates and modes are supported.

TASCAM HS-P82

Tascam's 8-track solid-state HS-P82 has eight microphone inputs (plus an internal slate microphone for naming takes), SMPTE timecode in and out, video and word sync, and a color touchscreen interface. The standard XLR microphone inputs include phantom power and analog limiting, with mic and line-level trims

controlled from recessed front-panel controls. A set of AES/EBU connectors is available for attaching pedigree A/D converters and preamplifiers. In addition, the HS-P82 can record a stereo mixdown for instant use. Audio is recorded at up to 192kHz/24-bit WAV format to a pair of Compact Flash cards, and users can record to both cards simultaneously. The Broadcast WAV files include iXML metadata for quick import into nearly any video or audio editing system, either via the USB 2 connection or a standard card reader. The HS-P82 further includes mid/side decoding.

ZAXCOM MAXX

The Zax-Maxx offers six analog inputs: four with mic preamps featuring NeverClip—which uses A/D converters per transmitter or recorder to generate 123dB of dynamic range—and two AES digital inputs that can be used with AES42 microphones. Each input channel has a highpass filter, two variable notch filters, variable delay and a soft-knee compressor. All inputs can be mixed into a 2-channel output bus over two sets of output connectors that can be individually configured as line, mic and consumer output levels, as well as a balanced mono and tape out. There are four AES direct outs and an AES stereo mix. The 2-channel balanced camera returns double as balanced line level inputs. Maxx comes standard with the ability to record eight tracks on a Compact-Flash card. Files can be recorded at up to 24-bit/192 kHz with a dynamic range of 137 dB.

ZOOM H6 HANDY RECORDER

Zoom's flagship H6 6-track recorder records WAV and MP3 files directly to SD cards and supports SDHC and SDXC cards with up to 128 GB. It can record up to six monophonic WAV files (or up to three stereo files) in up to 24-bit/96kHz formats. All WAV files are Broadcast Wave Format (BWF) compliant. The H6 has four main inputs (1-4) plus two additional inputs (L, R) that are derived from five interchangeable capsules: X/Y, M/S, shotgun or dual XLR/TRS combo. The XYH-6 X/Y capsule also provides a secondary input for connecting an external mic- or line-level signal via a stereo ½-inch Mic/Line In mini phone jack. Inputs 1-4 are combo connectors that can accept either XLR or ¼-inch balanced or unbalanced phone cables, and can handle both mic- and line-level signals. Each input has a dedicated gain control knob as well as a -20dB pad. [Editor's Note—Zoom recently introduced the H5; for information, go to New Products on page 66.] ■

olin, bass] and three singers." Both Price and Eisenberg also played hand percussion here and there, and Eisenberg also played a second fiddle. If it was, in fact, cut to 8-track, that would leave an extra track for the very few overdubs that were added, including a string section on three tunes and a very judiciously placed autoharp in spots on "I Scare Myself."

Botnick, whose father was a violinist with the Ted Fio Rito Orchestra, says he particularly liked recording Sid Page, whose hypnotic, mind-bending, gypsy violin solo is the centerpiece of the five-and-a-half-minute "I Scare Myself." He used a Sony C-37 on Page, either a C-37 or a Neumann U 47 to capture the guitars (Girton believes he played either an Epiphone or Gibson hollow body), and "probably a U 47 for the vocals," Botnick says.

Studio One was a famously good-sounding room, but the ambience on most tracks recorded there was augmented by what Botnick calls "a world-renowned live chamber. We would have used that in conjunction with an EMT [plate reverb] that I had worked on and gotten to the point where I could almost get the tonality of the chamber; I got the decay so it pretty well tracked [the chamber]." LiPuma comments, "It was a magic room with a magic echo chamber."

LiPuma is also quick to note that, "We weren't going to go for an older sound. The last thing any of us wanted was to make it sound like a period piece. They had a unique style all their own. It was reminiscent of a lot of things from other periods—Django Reinhardt and so forth—but

overall they had their own style; you knew it was them immediately. The singers weren't trying to sound like the Andrews Sisters."

"They were a lot of fun," Botnick adds. "Dan ran a tight ship with his people. They came in well rehearsed and ready to go, so the sessions were pretty quick, if I remember correctly. Tommy is really one for performance; getting the notes right, to be sure, but getting the emotion of the performance, which is my thing, too. I learned a lot about that from him. Being able to do that band live in the studio was wonderful."

Several songs on *Striking It Rich* got FM airplay when the album was released in the spring of '72, including "You Gotta Believe," "Walkin' One and Only" and "I Scare Myself." The band once again hit the road and this time also found themselves in their distinctive, cool, retro getups playing for national TV audiences on *The Tonight Show* (with Johnny Carson) and *The Flip Wilson Show*.

"That second [Blue Thumb] album was an absolutely magical time," LiPuma says. "There was something about all of them that you couldn't help but love. They were getting more and more popular. I would say they were an inch or so from breaking. But after a while there were problems going on in the band and between them and Dan. I think Dan got tired of dealing with everyone's problems."

Indeed, by the time they returned to make their third album with LiPuma, in 1973, Hicks had already decided to break up the band: "He had already titled it *Last Train to Hicksville*," LiPuma says. And though the sessions were more difficult, they still produced a fine album together.

Hicks and LiPuma—who went on to a storied production career that includes multiple Grammys—have remained close through the years; they used to be very close: "Sometimes when Dan came to L.A. he'd stay with my wife and me," LiPuma says. "At the time this [album] was made, we had one daughter who was 3 or 4, and when it was close to the point when our second child was going to be born, if it was a boy, it was going to be 'Dan,' and it was a girl it was going to be 'Danielle.' So my second daughter's name is Danielle. The great part is, every time Dan would call me after she was born, he'd refer to her by saying 'How's what's-her-name?'"

Hicks has had numerous bands since the original heyday of the Hot Licks, and he continues to regularly perform "I Scare Myself," still one of his most popular tunes. Thomas Dolby had a minor hit with a smoky, jazz-flavored version of the song in 1984, "and Muzak did a version, too," Hicks says with a mix of pride and amusement. "If you go on iTunes there a few versions by people I've never heard of." A splendid live version of "I Scare Myself" also appeared on Hicks' live, all-star 70th birthday celebration CD and video, called *Live at Davies* (after SF's symphony hall) recorded in 2006. Page, Price, Eisenberg and Leopold were all there, along with such friends and admirers as Rickie Lee Jones, Ray Benson, Tuck & Patti, Maria Muldaur and others. A fitting fête for a true original who's still bringin' it after all these years. ∎

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Taking a break from tracking, Maná poses for a quick snapshot in Studio E. Kneeling in front is drummer Alex Gonzalez, seated at right with acoustic guitar is vocalist Fher Olvera, seated at left is engineer Thom Russo, rear left is guitarist Sergio Vallin, and rear right is co-producer Jorge Noriega.

TIMBALAND, LIL WAYNE, AUSTIN MAHONE AND MORE AT THE HIT FACTORY CRITERIA MIAMI

In Studio A at Miami's Hit Factory Criteria, multiplatinum hitmaker Timbaland continued production on the soundtrack for the new drama series *Empire*, from Imagine Television in association with 20th Century Fox Television. Academy Award-nominee Lee Daniels and Emmy Award-winner Danny Strong created the series. Timbaland also wrote the original music, and Chris Godbey assisted by Perry Jimenez engineered the sessions.

In between tour dates, teen heartthrob Austin Mahone found time to visit the studio to work on some new material. He spent a few days in the 9000J-equipped Studio D recording vocals with songwriter/producer/vocal scientist Mischke (Michael Jackson, Jennifer Lopez, Trey Songz). The sessions were engineered by Ian Mercel and assisted by Sam Allison.

As preparations wind down for his upcoming tour with Drake, Lil Wayne camped out at the studio putting the final touches on his eagerly awaited next release. Various producers joined Wayne and mixing engineer Fabian Marasciullo in the studio's vintage Neve 8078-equipped Studio C. Andy Rodriguez and Sam Bohl assisted the sessions.

On the other side of the building in Studio F, Cash Money co-CEO Bryan "Birdman" Williams continued working with the CMR stable of talent. Engineer Fareed Salamah and assistant Sean Buchanan manned the SSL Duality for the sessions.

Returning to the site of their previous platinum successes, rock act Maná set up shop at the studio to record their as-yet-untitled CD. The band worked with co-producer Jorge Noriega and longtime engineer Thom Russo. Hit Factory Criteria staffer Femio Hernández also joined them for the festivities.

Studio Center Celebrates 42 Years and Number-One Studio Status

Studio Center is celebrating its 42-year anniversary, and in August, it celebrated its five-year anniversary of the new owners and management team.

"We pride ourselves in not only having great studios, engineers and producers, but also having great service," says Hector Mendez, president of Studio Center. "Since I came to the company, one of my biggest focuses has been on the service, which I think sometimes gets lost in this industry. I truly feel that our service is one of the key components to our success and continued growth in this ever-changing industry.

Additionally, the *Miami New Times* recently named Studio Center the Number-One Studio in South Florida.

Artists who have visited the studio recently include Chino & Nacho, who worked on "Tu Me Quemas" with engineer Jorge Taveras; Fonseca,
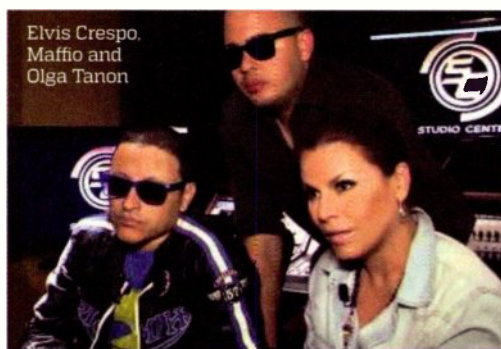
Elvis Crespo, Maffio and Olga Tanon

who worked on *Eres Mi Sueno* (which won a Grammy) with engineer Gary Bannister; Olga Tanon, who worked on *Una Mujer* with engineers Victor Mercader and Taveras; Elvis Crespo, who worked on *One Flag* with engineers Taveras and Mercader; Gloria Trevi, who worked on *De Pelicula* with engineer Taveras; Gente de Zona, who worked on "Ella Me Dijo" with engineers Taveras and Mercader; Henry Santos, who worked on "La Vida" with engineer Mercader; and Maffio, who worked on *Mi Sonido* with engineers Bannister, Taveras and Mercader. Maffio also produced all of the aforementioned projects.

Additionally, classical music artist Kento Masuda worked on *Loved One* with engineers Gary Vandy and Richard Maloney and producer Gary Vandy Audio Productions (Masuda also produced); Mike Dillon worked on *Band of Outsiders* with Vandy and Maloney engineering, and Dillon, Vandy and Maloney producing; HiTide worked on new material with Vandy and Maloney engineering, and HiTide and Gary Vandy Audio Productions producing; and Ignacio "Nacho" Nunez worked on his first solo project with Vandy and Maloney engineering, and Nunez and Gary Vandy Audio Productions producing.

MIXES THAT TRANSLATE
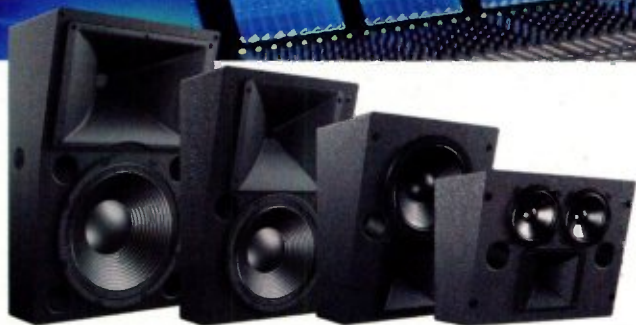
HMS-15   **HMS-12**   **HMS-10**   **HMS-5**

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DUNGEON RECORDING STUDIO

Oriente in Dungeon Recording Studio

Gospel singer/contestant on *The Voice* Donna Allen worked on new material with producers Jeffrey Davis and Jody Hill (Allen also produced) and engineer Fredric Freeman...Jazz group Oriente worked with producer Eddy Balzola and engineer Freeman...Peruvian rock band Campo de Almas worked with producer Jeffry Fischman and engineer Duane Baron...Rock band Romulus was with producers Henry Sosa, Daniel Fuenmayor, and Freeman (Freeman also engineered)...Haitian jazz artist Bemol Telfort worked with Freeman, who produced and engineered new material (Telfort also produced).

RED ROCK STUDIOS

L to R: Clay Ostwald, Carlomagno Araya, Cecile McLorin Salvant, Federico Britos

The studio's Yamaha C7 has become the favorite of Miami pianists, according to Clay Ostwald, owner and founder of Red Rock and award-winning producer and keyboardist for Gloria Estefan & Miami Sound Machine since 1986. There has been steady growth in acoustic jazz and classical recordings at the studio, which features Nuendo, Cubase, Logic and Pro Tools systems...Singer Khristian Dentley was in Studio A with engineer Clay Ostwald tracking new material for Take 6 in Studio A (Dentley also produced)... Jazz singer Cecile McLorin Salvant recorded a vocal session for Federico Britos' CD in Studio A with Carlomagno Araya producing and Ostwald engineering... The Henry Mancini Institute String Quartet was in Studio A with Stephen James Guerra, Jr. producing and Ostwald engineering...Brazilian pop group P9 was in Studio B working on a pre-production session, with Keithin "Mizzle" Jamal Pittman producing and Dorian Reyes engineering...Trinity McCray was in Studio B working on pre-production for *Total Divas* music, with Reyes producing and engineering...Timmy Thomas and Alana Joy recorded a remake of Thomas' classic hit "Why Can't We Live Together" in Studio B, with Reyes producing and engineering...Lindsey Blair Quartet tracked and mixed *A New Dawn* in Studio A, with Blair and Ostwald producing and Ostwald engineering.

NOISEMATCH STUDIOS

Farruko

Pop singer Claudia Ray worked on "Alive" with Alex J. producing and engineering...Alex J. tracked Latin Grammy-nominee and urban/reggaeton artist Farruko, Latin Grammy-nominee and reggaeton singer J. Balvin, and Mexican pop band La Santa Cecilia for Terra Live Music sessions...Jazz-fusion group Aaron Lebos Reality tracked guitars, bass, keys and drums with Alex J. engineering and the band self-producing...Latin pop artist Alex Miró worked on his *Elemental* EP, with Alex J. producing and engineering.

THE PANIC ROOM STUDIOS

Cumbia group Kumbia All Starz, Grammy-nominated pop singer Cristian Castro, Mexican pop singer Verónica Castro, bandleader/salsa and bolero singer Gilberto Santa Rosa, singer-songwriter Jon Secada, guitarist/singer/composer José Feliciano, and Marciano Cantero (lead singer/bassist for Los Enanitos Verdes) were all in the studio recently working on material, with AB Quintanilla and studio owner Luigi Giraldo producing, and Giraldo also engineering...Producers Yarka Miller and Giraldo worked on *Navidad Con Amigos* for the Telemundo Network and *Nuestra Belleza Latina* for Univision Networks (Giraldo also engineered both projects)...Singer-songwriter/musician Jencarlos Canela worked on material with producers Julka Brothers and Julio Reyes and engineer Andres Bermudez.

Velvet Basement Studio Acquires Historic Console, Relocates

Robert Dante

Velvet Basement Studios has recently relocated into a 1,200-square-foot facility and acquired an SSL 6000 console, which was formerly owned by the late, great Tonight Show legend Johnny Carson.

"We had been using the flagship Yamaha DM2000 digital console for all of our mixing for the last decade, and it has treated us amazingly," says Robert Dante, studio CEO and founder/chief producer and engineer. "The DM2000 is quite a solid piece of gear. But digital is digital and analog is still what we like to hear. So we have finally done away with the digital console, and the old space altogether, and now have the SSL 6000 console, which was formerly owned by Johnny Carson! We love the sound of this analog beast. Its warm and punchy character has been a nice addition to our overall sound. Sure it was great having total recall with flying faders using the DM2000, but we don't think about those fancy details very much once we heard what this thing is capable of."

The control room is an acoustically sound, floated-floor room in a room, with hardwood floors and ample bass traps and acoustic treatment, which Dante says provides "a very tight bottom and very accurate listening environment."

Also noteworthy is the studio's 500-square-foot live room with hardwood floors, which is equipped with a new six-piece set of Premiere drums and a new Kawai acoustic piano.

In Velvet Basement's quest for "more analog," the studio also just added a Manley Vari-Mu Compressor/Limiter to its collection of gear. "We love the way it glues together our mixes, delivering a sweet, smooth flavor rather than a transparent one," Dante says.

Velvet Basement also has a lounge with a kitchen, leather couch, flat-screen TV, and an original vintage Galaga arcade game from the '80s, which, according to Dante, "our clients love to jump on when they feel need to shoot some aliens."

Recent artists who have visited the studio include international preacher/motivational speaker/author Dr. Mattie Nottage, who worked on a Christian/Gospel single for her ministry; writer/guitarist Michael Burmin, who is working on songs for artist; PinoGrillo, who is recording a pop/urban album; and Albert Camino and his band Analog, who recorded their indie-rock album. All projects were engineered by Dante.
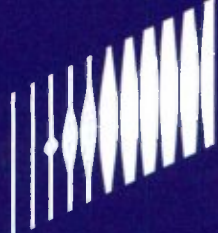
MARCELLA ARAICA
MAKING WAVES IN MIAMI RECORDING

BY BLAIR JACKSON

Even as women continue to make great strides in areas from politics to corporate boardrooms, they are still rare in the recording business outside of studio manager and clerical positions. When Miami mixer/engineer/composer Marcella Araica attended the Full Sail Production and Recording Program in her home state around the turn of the millennium, "I had a class of around 165 or 170, and only five of us were women," she says. "That was my first taste of seeing women being such a minority in the field. Out in the world, women are thought of us being teachers and nurses and doing blue-collar work, which I found eye-opening. After I graduated and started getting into my career, I'd talk to younger girls about what I did, they all said, 'I didn't know I could do that,' and it's because women are not educated to know what they can do. I think a lot of women are still taught to go to school and eventually become mothers, have our children and that's the end of the story. But I think there's an opportunity for women to do all the things that they put their minds to, do them as well as men do, and also achieve what we were put on this earth to do, which is to bear and raise children."

She should know. Not only has Araica (pronounced "ah-rye-ka") established herself in the 12 years she's been out of Full Sail as a top engineer, mixer and composer—with a glittering client list that includes Timbaland, Missy Elliott, Britney Spears, Madonna, Pink, Duran Duran, Usher, Justin Timberlake and Nelly Furtado, among others—a few months ago she also became a mother for the first time. When I ask her about whether she was able to take a real maternity leave, she laughs. "I had my son [Kayden] in Naples, Italy, so the first month I was still in Italy, but then as soon as I got

back to the States, I had to finish building my new studio in Miami, and at the same exact time, my artist, Luke James, who I signed to my label, NARS Records, his album had to be turned in because it releases [in September]. So I didn't get the time off that I really wanted, but it's working out."

When Araica started at Full Sail, she wasn't sure what path her career might take. "I loved what [producer] Linda Perry was doing," she says, "and I think I was trying to follow her a little—the songwriting aspect and the musicianship—but I ended up going headfirst into the engineering end of it." It turns out she was a *very* fast learner, and her rapid advancement in the field did not sit well with some of her peers.

"I got a hard time from some people who didn't like that I moved up so quickly," she reflects. "I remember when I started my internship [at Hit Factory Criteria in Miami], I got my first assisting gig two months after I started, and there were other interns there who had been there nine months and never worked a session. So I got a lot of heat, a lot of hatred. There was another assistant, too, who just did not like the fact that I was a woman doing it, and he'd come at me like, 'Are you sure you can pick that amp up? It's so heavy.' I got tired of that.

"But I definitely had the drive. I'll be honest—you go to Full Sail and you learn what you learn, which is a lot, but then it's a different thing when you get put out there in the lion's den. It's like, okay, now there's no time or place to say, 'No,' there's no time or place to say, 'I can't do this.' You *do* it, you know? That's what I excelled in. Even if I wasn't comfortable doing something, I tried my hardest and I still made it believable. And if I blow up the room..." She laughs again. "I learned from all those experiences. I learned from patching things in the wrong way and feedback blaring through the speakers, loud pops, whatever. I looked stupid at times at my sessions, but you learn quickly—'I'll never do *that* again.'"

She counts Jimmy Douglass and Demacio "Demo" Castellon as early mentors in the field, and she lucked out that two of her first high-profile clients—Timbaland and Missy Elliott—also had knowledge to share and the patience to let her develop her chops on the job. Up to a point: "Missy used to call me a turtle in the beginning, because I was so slow at Pro Tools," she says with a chuckle. "She'd just kick me out of the chair: 'You too slow!' The only way I could become proficient enough was *not* in front of her, so at the same time I was working for her, I was taking little neighborhood sessions when I could fit them, just for the experience. That's partly where I learned to be fast, and learned quick keys and learned how to EQ on the spot, without having to think about it, without being embarrassed about it. I had to grow my confidence like that." And it ended up being Missy Elliott who gave Marcella the nickname she carries to this day—"Ms. Lago," after the speedy Lamborghini Murcielago sports car—because of her fleet-fingered Pro Tools work.

It helped, too, that she was smart, friendly and easy to work with. "The studio manager who hired me [at Hit Factory] originally and is still a dear friend, Trevor Fletcher, always said, 'Fifty percent of this job is technical and the other 50 percent is your personality.' The social game of it is a big part. And at the end of the day, it's not my gig. If I'm working for them, I have to do whatever it takes. You don't want to come off as a know-it-all: 'This is the way I do it, so I'm gonna do I this way.' You have to be open."

It was through Timbaland that Araica hooked up professionally with producer Nate "Danja" Hills. The two have worked on many projects together through the years and have also been studio partners for some time, including their brand-new facility in Hallandale (North Miami). "It's a three-room complex with a dedicated mix room, Danja's producer room and a small writing/vocal room," she says. "We have two SSL 9000s—the one in the mix room is a 72-channel SSL 9J, the producer's room has a 40-channel 9J and the other room has an SSL Matrix. So it's a supercool high-end facility, perfect for what we're doing." She does most of her mixing on good ol' Yamaha NS-10s ("I've been doing that since Day One"), though her mix room also has Augspurger TAD 1603s for mains.

In an age when so many artists and engineers have eschewed big consoles and analog outboard gear in favor of working exclusively in the box with plug-ins, Araica is happy working on the equipment she came up on and spending the time she feels she needs to get everything right. It's an old-school approach she fears is fading: "A lot of the newer artists don't understand that kind of work ethic and that's frustrating to me, because even though I've only been doing this for 12 years, I actually got into it at a time when I got to see that work ethic. Now everything is so fast, fast, fast, 'Let's just get it done!' It doesn't feel like there's as much appreciation of the art that's behind it. Like, I see mixers who take pride in doing two or three mixes in a night, and that's fine—you can do that. But they look at me and I'm still using 80 channels of the SSL and they're wondering, 'Why do you do that? Everything is in the box now.' Well, that's not how I work. I don't care if people call me a dinosaur. I like the analog sound. I like passing everything through the board. I like EQing

everything individually. I like that it takes me six to eight hours to do a song."

Which isn't to say she abhors plug-ins. "No, no, not at all," she says. "I do love those UA plug-ins. And plug-ins in general have come a long way. If it sounds good, I want to use it; I'm not opposed to it. But I still prefer the sound of raw analog gear. I *like* the noises some of the machines make. 'Oh, my god, we've got a little buzz!' Yeah, just leave it! It adds character. Not if it's obnoxious, of course, but if there's a little low hum, it might just be what the unit is bringing to the record. It is what it is, and sometimes it adds to the overall sound."

Though her work has taken her far and wide—including to London, where she worked with Madonna and Duran Duran—Araica still loves living and working in Miami, where she was born and raised. In addition to its enormous Latin music focus, the city has long had an extremely vibrant hip-hop/R&B scene, which Araica has been plugged into from the moment she got out of school.

"The recording community here is pretty small, nothing too crazy," she says. "It's definitely not like L.A. today, or New York the way it was before all the studios closed down. But Miami is a tight-knit community. It's almost like a small town. If you work in the music business down here, you probably know almost everybody. If you need a writer or a musician, there's a matrix out here and people help each other. I sometimes wish it was bigger—though not too big. But people love Miami, so they're always going to come here."

And there is plenty of work coming her way. In fact, it's a wonder she can juggle the myriad commitments she has between mixing, composing, engineering, her record company (NARS) and being a mom. How does she do it? "I like to pretend that there are 30 hours in a day; that's for one. I also have the help of an assistant; the rest is all in my head and my Blackberry. I create project lists and have charts up in my office and my studio and I always try to keep a mental note of remembering what's happening. I've been working at this pace and doing it so long now, I've figured out a natural way of going about it all. So far it's been working."

Any last words of wisdom? "My philosophy is you dream with your life and you love every minute of it. So if you see something you want to do, just go for it." ■

THE FUTURE IS NOW

By Gino Robair

We sacrifice a lot to be in the entertainment industry. At some point, most of us have weathered strained relationships with friends and lovers, gone for long periods of time with irregular sleep, and missed entire weeks of sunlight as if we were living at McMurdo Station in June. Yet we barely notice because we love what we do.

At some point, we *become* our job: We live it.

That's a good thing, right? Especially when we're in the early part of our careers and can devote everything to our work. But becoming our work doesn't mean that we are getting all we need from it, even during the best of times. As most of us eventually figure out, the more successful we are, the harder it is to get some of the things we really need to keep us going.

I'm not suggesting long-needed vacations or even time to meditate or getting physical exercise: The thought of that, alone, can stress many of us out. Rather, I'm talking about doing the things we actually want to do, not the things we think we have to do for someone else (or even the things we've been doing for so long we imagine it's all we want to do).

Somewhere lower down on our To Do lists are items that are very important to us for some other reason than income generation or career building, but we've put them aside for the very reason that they don't put bread on the table. They may even seem a little frivolous because of that. Often these projects are pushed so far down that they leave the To Do list altogether and become part of a Wish list—a collection of things we'll come back to at some point in the future "when there is time."

What brought this to mind was the editor's note in the September 2014 issue of *Drum!* magazine, where editor-in-chief Andy Doerschuk talks about losing the ability to play drums due to spinal stenosis, an exhaustingly painful condition. Although misdiagnosed initially as shin splints, the condition has worsened to the point where Doerschuk can no longer play his instrument.

"But what would you do if you couldn't play drums?" he asks in the article. "...it feels like losing one of the best friends you've ever had."

Anyone who has broken a limb or has had a repetitive stress injury suddenly realizes that even the most mundane chores—brushing your teeth, taking a shower, getting dressed—cannot be taken for granted. The pain of any disability is heavy enough to deal with, but it's a major blow to lose the one thing that feeds you to the core of your being:

> **Rather than suggest we take better care of ourselves or take more time off to recharge, I am going to suggest that we do more.**

"Drumming was my escape from life's challenges, my constant companion, a source of pride, my identity."

As I read Doerschuk's letter, I re-evaluated my own relationship to the drums, my main instrument, as well as other aspects of my life and work that are sources of pride, identity and escape mechanisms. The first stage in such an evaluation is to become mindful about the parts of your job that are the most meaningful to you; a reminder to enjoy yourself and not take things for granted.

But it's stage two of self-reflection that I want to focus on; a look at the important things that we have yet to accomplish and that we've always wanted to do but just haven't gotten around to. That's what I couldn't help thinking as I read Doerschuk's moving letter. It's the place where my mind goes when a friend or colleague is seriously injured or dies. That question of, "Why did I wait so long to..."

So rather than suggest we take better care of ourselves or take more time off to recharge, I am going to suggest that we do more.

Scan the lower parts of that never-ending list of chores and locate those projects that you've been putting off forever, the ones that will take you away from your comfortably insane workload. These are the things you don't want to remember with regret during the last moments of your life (or after someone else has passed) and wonder why you didn't do them.

Is it a collaboration with someone you admire, perhaps someone who suggested you work together? Or maybe you have always wanted to produce and release something of your own, though it requires some pro bono assistance to yourself for a change, rather than for someone else's cause.

We sacrifice a lot to stay busy and relevant and have no problem giving back to others. Why is it so difficult to give a little back to ourselves? We know we're not going to be here forever, nor will the people we enjoy working with the most. Why deny ourselves the chance to do the things we've been dreaming of but keep putting off for later.

Doerschuk finished his message by writing, "Don't hesitate. Get out and play drums however, whenever, and wherever possible, not for the money, but because you can."

I suggest his words be taken as a metaphor for how we can approach our own skills and loves within our work/life. Take a step back and do what you need to do for yourself, while you are still able to. Don't leave the most important projects for the future. ∎

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SE ELECTRONICS SPACE

Portable Acoustic Enclosure

sE Electronics has redesigned its Reflexion filter and named it SPACE ($399). It features a larger surface area, thicker layers of different acoustic materials for better absorption, deeper air gaps for better isolation and better diffusion, which helps to randomize primary reflections. The design features seven vertical pillars that significantly deepen the air gaps internal to the device, while simultaneously aiding in the randomization of diffusion. SPACE also boasts new hardware for clamping the device to any microphone stand or drum hardware. Fully adjustable and lockable vertically and horizontally, and with a tilt function, SPACE is stable and can be positioned accurately.

ANTELOPE AUDIO ZEN STUDIO

Little, Red I/O Powerhouse

The Zen Studio from Antelope Audio ($2,495) takes the audio interface to a new level of capability and portability. The single-rackspace unit features 12 Class-A mic preamps with phantom power, eight mic/line inputs, four mic/line/instrument inputs and two independently assignable headphone outputs. Its I/O includes two 25-way D-Sub connectors (eight channels of I/O), one stereo monitor out, four ADAT connectors (up to 16 channels of I/O), two S/PDIF RCA connectors (I/O), two inserts (TRS), two word clock BNC connectors (I/O) and one low-latency high-bandwidth USB 2 port. Zen Studio comes with routing and mixing software plus parallel DSP processing for cutting live tracks with plug-in effects.

AUGSPURGER DUO 12.3 DSP

Active Bigs

The Duo 12.3 DSP ($26,500) from George Augspurger offers the performance and sound of the original Augspurger design in a compact format, but with all the clarity, depth and power of its larger siblings. The main monitor cabinets offer dual 12-inch drivers and the distinctive Augspurger wooden horn incorporating custom compression driver with beryllium diaphragm. The subwoofer cabinets are separate with high-excursion 12-inch drivers, all powered by cabinet mounted Class-D amplifier systems with 2,500 watts per side (1,000W to sub; 1,000W to midrange drivers; 500W to horn). DSP provides the crossover, limiting, driver alignment and system equalization and is controlled through included software and a USB interface.

ZOOM H5 RECORDER

Affordable Handheld

Zoom's H5 ($269.99) features a system of interchangeable input capsules, operates up to 24-bit/96 kHz, has four inputs and two outputs, and operates as a USB audio interface. Other features include two XLR/TRS combo inputs, two mic preamps, and an X/Y microphone attachment. The integrated shock-mount reduces handling noise while the large backlit LCD display allows level monitoring, even in bright light. Controls include analog-style gain adjustment with "roll-bar" type covers and a Hold switch disables front panel buttons to prevent accidental operation during recording. Other features include Auto-Record, Pre-Record and Backup-Record functions and Loop Back operation for combining live audio input with computer playback.

MOTU 1248, 8M, 16A AUDIO INTERFACES

Thunderbolt, Mixing and Networking

MOTU has announced three new Thunderbolt audio interfaces ($1,495 each) with complementary I/O configurations, A/D/A conversion with very high dynamic range, 48-channel mixing, DSP effects and AVB Ethernet audio networking for system expansion. The 1248, 8M and 16A differ only in their analog I/O configurations. The flagship 1248 offers 8x12 balanced TRS analog I/O, four mic inputs with digitally-controlled individual preamps, two front-panel hi-Z guitar inputs, two independent phone outs and stereo RCA S/PDIF digital I/O. The 8M provides eight balanced TRS analog outputs, plus eight mic/line/instrument "combo" style inputs individually equipped with digitally controlled preamps, 48V phantom power, pad and MOTU's hardware-based V-Limit overload protection. The 16A is packed with 32 balanced TRS analog connections (16 inputs and 16 outputs). All three units provide two banks of optical digital I/O, word clock I/O and computer connectivity through either audio class compliant USB 2 or Thunderbolt (1 and 2 compatible).

ICON UTRACK SATELLITE PRO INTERFACE

I/O for Live and Desktop Applications

The Utrack Satellite Pro from Icon ($219.99) operates at up to a 192kHz sample rate and offers a USB port, two analog line inputs (TRS), one mic input with phantom power and a DI input. There are also two headphone outs with separate gain control, two line outputs and Icon's Virtual Port technology allowing internal computer audio sources to be routed independently to or from any input or output. The unit supports DirectSound, WDM and ASIO 2 and is compatible with Mac OS (Intel Mac) and Windows XP, Vista (32-bit/64-bit), Windows 7 (32-bit/64-bit) and Windows 8 (32-bit/64-bit).

ULTIMATE EARS 900 IN-EAR MONITORS

High-End Portable Model

The new 900 monitors from Ultimate Ears ($399.99) promise an excellent fit and listening experience. Features include four proprietary precision balanced armatures, sensitivity at 101.2 dB (50mW, 1 kHz), and 26dB reduction of outside ambient noise. Other features include a 3-way crossover and patented dual-bore audio paths to keep signals separate, which promises unparalleled clarity. Its 48-inch cable terminates with a 1/8-inch gold plated connector. The 900 includes nine pairs of specially designed ear cushions, six sizes of silicon cushions and three sizes of memory foam cushions, each designed with a large-bore opening to assure a perfect fit.

MESANOVIC MODEL 2 MICROPHONE

Motor City Dynamic

The Model 2 ($649.99 each; $1,249.99 stereo pair) is a passive ribbon mic that is completely handcrafted in Detroit. It features the same ribbon dimension found in the company's Model 1, but with some alterations. The Model 2's ribbon motor features a shorter path length from front to back than Mesanovic's Model 1, as well as custom-designed resonator plates. These features are said to give the Model 2 an extended high-frequency response while retaining a classic ribbon midrange and promising a rich low end that reaches flat out to 20 Hz. The Model 2 also features a custom toroidal transformer from Samar Audio Design that provides low distortion, low noise and high-frequency detail. The microphone's slim 1.25-inch body allows it to be used with most third-party shock-mounts. The Model 2 is designed for capturing vocals, guitar amps, drums, strings, brass, piano and more. Mesanovic sells the Model 2 directly from its Website, and offers a lifetime warranty. The microphone ships with a plastic molded case, shock-mount and protective pouch.

New Sound Reinforcement Products
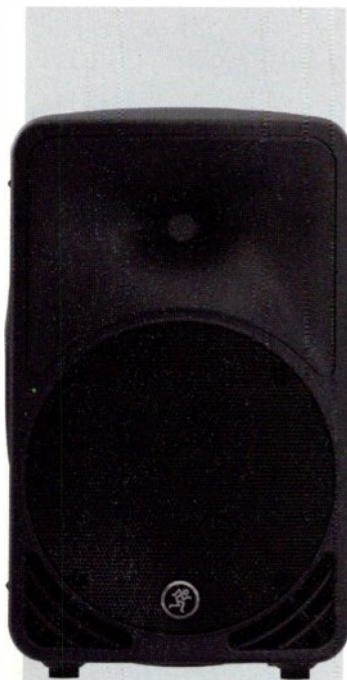
MACKIE SRM SERIES LOUD-SPEAKERS

New Features, New Price

Mackie has redesigned its two-way molded enclosure speakers, which now offer a larger 1,000-watt amp. DSP sound shaping and a lower price. The SRM450 ($629.99) and SRM350 ($519.99) feature onboard processing including four speaker modes (P.A., DJ, Monitor, and Solo), an intelligent, single-button feedback destroyer with 1/16th-octave filters, Smart Protect DSP and an integrated 2-channel mixer.

YAMAHA CL CONSOLE VERSION 2 SOFTWARE

Upgrade Brings Big Changes

Yamaha Commercial Audio Systems has released Version 2 software for Yamaha CL Series digital audio consoles, available as a free download. New features include discovery and head amp control for the new, compact QL Series, and DCA Roll-Out, in which channels assigned to DCA groups can be instantly called up to the console faders for enhanced operational flexibility. Mix Minus, an important feature in broadcast applications, is now provided; with one simple operation the signal from a particular channel can be removed from a specified bus. With V. 2, it is now possible to create read-only scene memories. A new Daisy Chain Insert feature allows two devices to be inserted into one channel or bus for enhanced processing freedom. A GR Meter (dynamics meter) Option within the Channel Name Display will show the Dynamics 1 and Dynamics 2 gain reduction meters in the channel name display.

CYMATIC AUDIO UTRACK 24 RECORDER

Capture, Playback and I/O

The Cymatic Audio uTrack 24 Recorder ($999) allows connection of 24 channels of balanced input/output through 25-pin D-Sub connectors and 24x24 I/O using the optional digital expansion module. The unit operates at a 24-bit/96kHz sample rate and records directly to a USB hard drive attached to a front panel USB port. Features include 3-segment hardware input metering per input channel, large, easy to use illuminated front panel transport controls, and an internal DSP mixer allowing internal monitoring of a stereo mix of all 24 inputs through a front panel headphone output. Users may adjust level, panning, mute and solo for all 24 inputs while using 24-segment hardware input metering, all with no external mixer required. Two uTrack 24 units may be synched for 48-channel record/playback, and the unit also works as a 24x24 computer audio interface.

ALLEN & HEATH QU UPGRADE

Firmware Brings New Features

The Version 1.5 upgrade for Allen & Heath's Qu series consoles (free download) offers a range of new features and performance enhancements including the addition of four DCA Groups, channel naming and name synching to iPads and Allen & Heath's ME-1 Personal Mixer, flexible output patching on Allen & Heath's AudioRacks running the dSNAKE protocol and monitor sends to ME-1, and support for Allen & Heath's Qu-32. USB recording sources can now be set to Insert Send or Direct Out. MIDI Machine Control has been added to the MIDI screen and SoftKeys functions, and V 1.5 now supports the company's AB168 AudioRack.

ADAM
PROFESSIONAL AUDIO
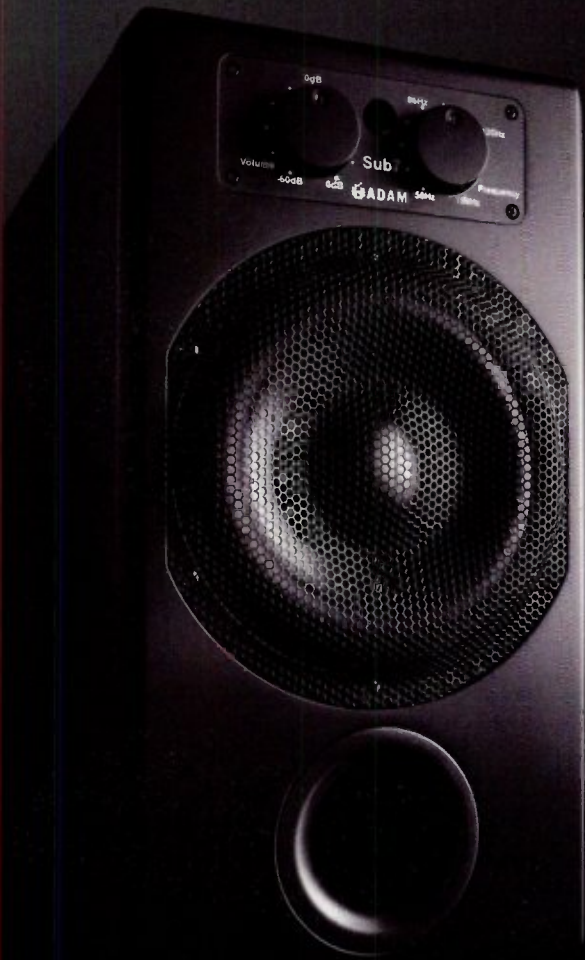
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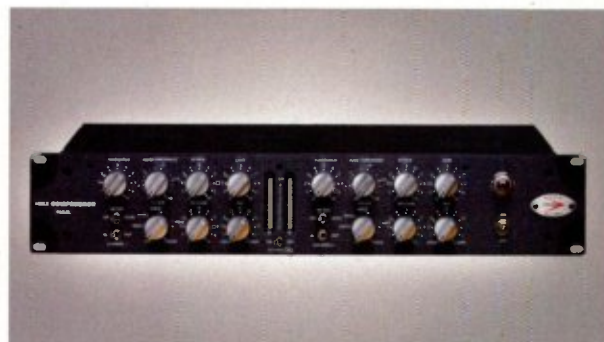Promo price: $1249

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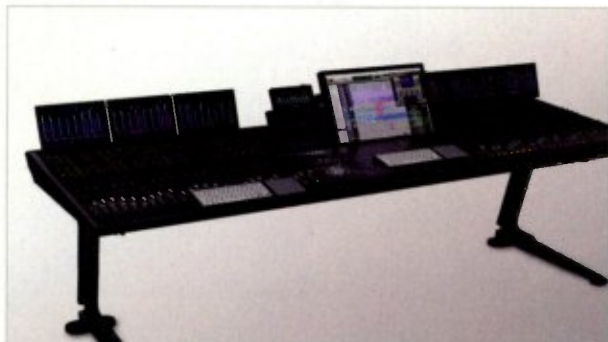
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Mix faster and easier with intelligently designed controls, touchscreen, and intuitive modular layout

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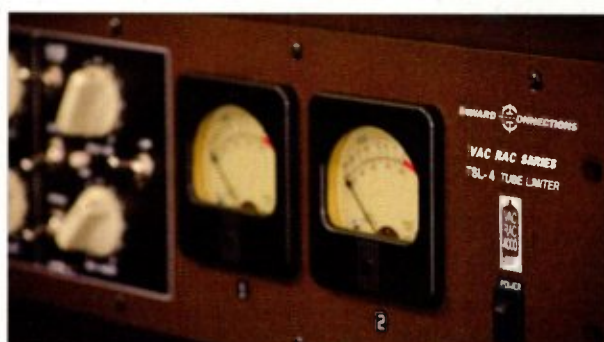
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PMC TWOTWO™
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NEUMANN TLM 107
Far from evoking vintage or retro nostalgia, the standard-setting TLM 107 represents an impressive, modern studio mic
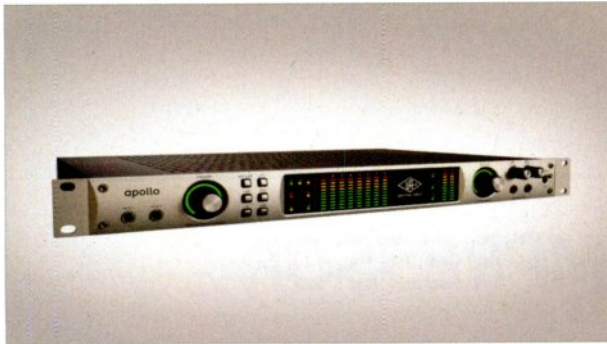
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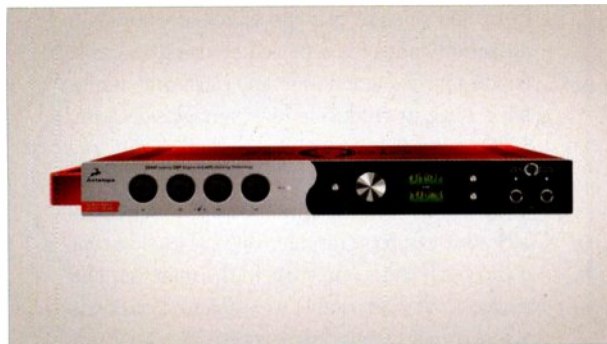
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By Wes Maebe

Tech // reviews

ANALOGUE TUBE AT-1 COMPRESSOR
New/Old School Fairchild-Style Limiter

The AT-1 uses 12 tubes: 6386LG, 6084, 5651, 12AX7, 12BH7, 6BL7, 6L6GTA, and GZ34.

I first bumped into Simon Saywood at an AES in San Francisco when he was showing off his then-brand-new Fairchild 670 copy, the AT-101. It became instantly clear that Simon was not into this just to make yet another copy of a classic. Being one of London's top maintenance engineers, Saywood takes great care and pride in researching how the original units behaved and which components were responsible for the sonic character. In his designs, he sticks as close to the original specifications as possible, yet improves where possible and appropriate. I test-drove the AT-101 when it came out on an Elliott Randall session. Needless to say, I couldn't wait to hear how the AT-1 was going to compare and what it would bring to that infamous Fender Strat.

FIRST GLANCE
The AT-1 is impressive. It is 19 inches wide, six rack units tall, and weighs 44 pounds—this is not a small unit. The front panel is a lovely brushed matte black with the same control knobs as the stereo AT-101, but in mono. It gives the user control over input level, threshold, attack and release times, bypass and external input. And, of course, it wouldn't be a true Saywood Analogue Tube without one big mother of a VU meter.

It is recommended that the unit be kept more or less freestanding in order to provide airflow. Taking a peek around the back will make this very clear. The AT-1 has 12 tubes poking out of the back (6386LG, 6084, 5651, 12AX7, 12BH7, 6BL7, 6L6GTA, and GZ34). Drawing around 110W of power, the unit gets quite hot.

GEEK SPEAK
The AT-1 was designed and developed to perform and operate like the Fairchild 660. Gain reduction control is achieved by using a double triode gain reduction amp and push-pull amplifying stage to produce a high-voltage sidechain. This design makes for a very natural sound, low distortion and no audible thumps or pops, even under severe limiting conditions. The AT-1's attack time can be set at microsecond level so that it can catch short transients in the material. The release time is adjustable from 0.3 to 25 seconds across six click stops. Two of these six release settings have automatic functions dependent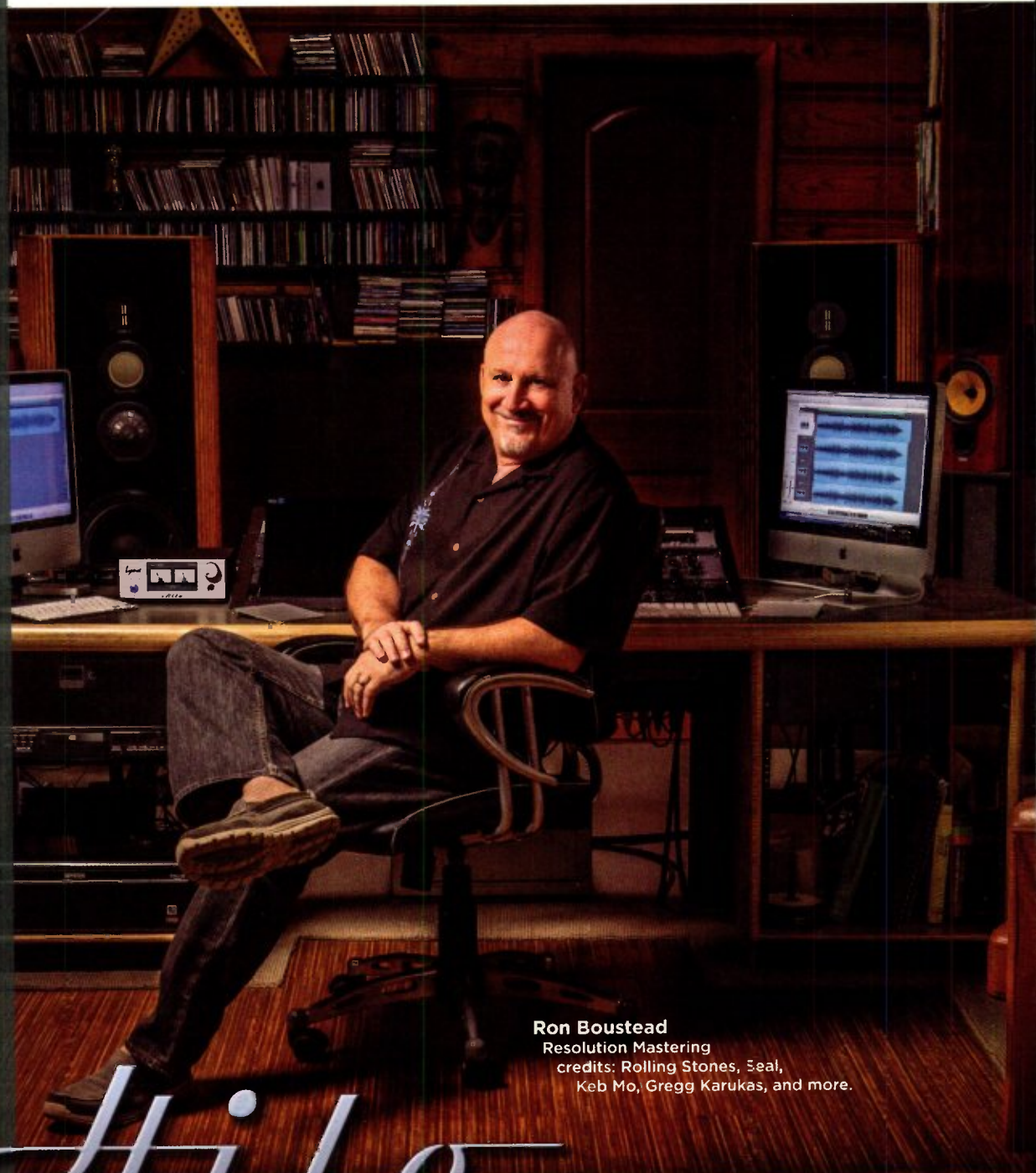 on the program material, offering fast recovery for quick peaks and longer recovery times for high-level material. All the Time Constant values are exactly the same as on the original 660 and 670 units. Like its 660 ancestor, the AT-1 uses a single independent feedback limiter and was designed for use in regular line-level circuits, like the mix bus and channel inserts.

This unit is not just half of the stereo AT-101. The mono unit has its own character and, like the 660, sounds and behaves differently compared to its stereo counterpart. A few of the main distinguishing features that set it apart from the stereo unit are the valve PSU and attenuator control. It is a tapered step attenuator, remade exactly the way it was back in the '50s. Some of the new features are the stepped AC Threshold control, making it easy to recall, and when you're stereo linking two units (or more for surround), this control boasts a neat one-percent accuracy per step. The AT-1 also comes with an external key input, which you can easily engage with the key switch. This splits up the gain reduction and AC threshold circuitry, allowing you to access the com-

pressor's sidechain when you require de-essing or ducking.

The AT-1's attack times range from .2 to .8 milliseconds and the release positions give you the choice of 0.3, 0.8, 2 and 5 seconds, in addition to the two automatic function program settings. Depending on the DC threshold setting, the AT-1 will provide ratios from 1:1 to 1:20. In addition to the original tapered step attenuator for the input gain and the time constant switches, it has a 21-step AC threshold, a continuous DC threshold control, the bypass that switches the AT-1 completely out of circuit, the bright green external input switch, stereo link and the metering switches. The inputs and outputs are all transformer-balanced and on XLR connectors wired with pin 2 hot for Europe and pin 3 hot for the U.S.

IN THE STUDIO

I wanted to hear this unit on Elliott Randall's Strat and hear how it compared to the stereo version. I put up a dynamic and a ribbon on the cab and strapped the AT-1 across the dynamic channel, straight to disk. The mic went from an ultra-clean preamp into the compressor and into the DAW. No electronic detours. We both know the amp and the mic really well. The resulting sound was humongous. Generally, I blend a little of the ribbon in to supply some warmth and roundness. What happened here was that the dynamic combined with the AT-1 was perfect. What's more, it did not kill the use of the ribbon—it actually made the ribbon and the dynamic signals gel together even better. I decided to drive the input stage quite hard and had it compressing very gently. The AT-1 smoothed out any little level

jumps, and Elliott's signature volume swells just leapt out and sang.

Just before we hit the red button, I did one of my favorite things with the Analogue Tube units. I hit the bypass and had the artist play, then slung the unit into circuit, unannounced. Seeing an artist's face light up and get inspired by the sound they're hearing must be the biggest compliment to the unit and its designer. Well, that and the exclamation: "Oh my God, I want one!" (That would be me shouting that.)

By now I was seriously vibe-ing on the AT-1 and wanted to hear more. I subsequently tried it on various vocals—high and screaming female vocals, funky male vocals and really dark, husky male vocals. All of them blossomed, and hearing the results back in the headphone mix made the singers perform significantly better.

Finally, I had a bass in an EDM track that was flatter than a pancake. The source sound wasn't very exciting. The moment I hit the insert and had the AT-1 across it, with plenty of gain driving those tubes, that bass came out bigger than a house. The track started pumping, and all the bottom end of the song just glued together.

This is an excellent replica of the original Fairchild 660 in shape and function, and it makes me think back to tape. It looks like an original, it sounds like an original, it's cheaper than buying an original today, and it'll require a lot less maintenance. Every studio should have one. ■

Wes Maebe is a UK-based recording, mixing, mastering, and live sound engineer. You can check out his work at wesonator.co.uk.

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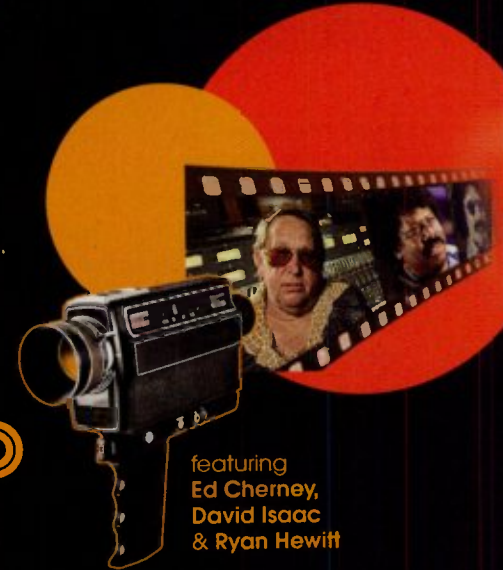Audio Trades-Man

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UNIVERSAL AUDIO

Tech // reviews

SAMSON RESOLV RXA5 AND RXA6 MONITORS

Ribbon Tweeter, Rear Port, HF/LF Controls
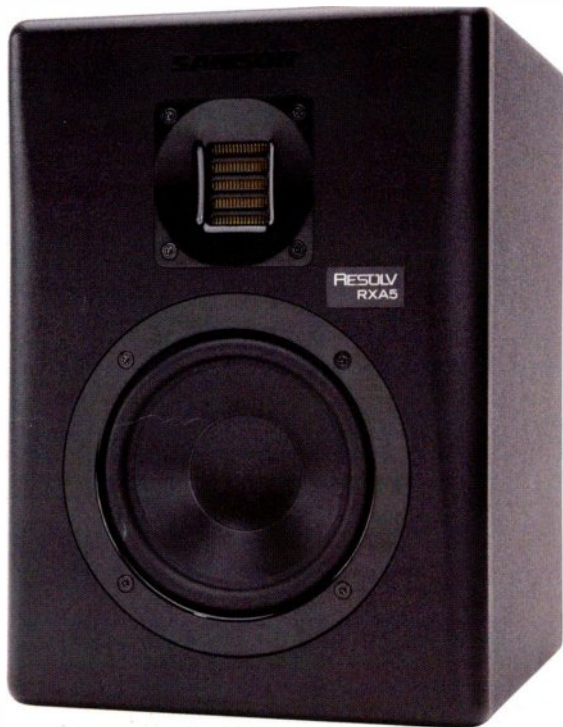
It seems that every month a new studio monitor hits stores. Companies with many years in the game, like JBL, Tannoy and Genelec, are constantly facing competition from new upstarts. So how can any product cut through the competition and allow itself to be heard? Do something different. Samson has been making the Resolv line since 2000 including the Resolv, Resolv 50, Resolv 65A, ResolvA, and the Resolv SE. The first line of Resolv monitors came in three varieties having a 5-, 6- or 8-inch woofer, each with a 1-inch soft-dome tweeter. The latest Reslovs, the RXAs, have found a great way to grab attention by incorporating a ribbon tweeter.

Ribbon tweeters are surprisingly rare in studio monitors, with ADAM leading the market and few others following. Every engineer has an opinion about ribbon tweeters, but they are significantly more efficient than dome tweeters when it comes to high-frequency reproduction. A large surface area is folded into a smaller space and compresses and expands perpendicularly to the listening position. According to manufacturers, this allows them to move air four times more efficiently than dome tweeters, resulting in crisper transients and, theoretically, less ear fatigue.

COMPONENTS

The RXAs come in two models: the RXA5 with 5-inch woofer and the RXA6 with a 6-inch. The woofer material is a copolymer with the surround made of butyl rubber, chosen for its dampening properties, providing clean, tight bottom end with low distortion. The "Air Displacement Ribbon Tweeter" is made from 2.5 inches of folded aluminum. As in most monitors today, the tweeters sit behind a sculpted waveguide designed to disperse sound, widening the listening field while maintaining a clear stereo image. The RXAs feature 24dB/octave active Linkwitz-Riley crossovers centered at 3 kHz on the RXA5 and 3.5 kHz on the RXA6. Each model features a bi-amped design, with the RXA5 delivering 20 watts to the tweeter and 50W to the woofer, and the RXA6 feeding 25W to the tweeter and 75W to the woofer.

On the back panel there is an overall level control, using a raised, rubber-capped knob. There is a single detent at the center position. While there are no individual controls for the high- and low-frequency amplifiers, there are recessed controls for high- and low- frequency shelving, which can be adjusted with a flathead screwdriver or your fingernail.. Each is detented at the flat center position and offers ±6dB of boost and cut. All three standard analog connectors are offered: unbalanced RCA, balanced ¼-inch TRS, and balanced XLR. There are no connectors for stereo linking. A fuse, a rocker-style power switch, and universal power connector round out the back panel.

The enclosures are made of matte black vinyl-wrapped MDF. The edges of the front side are sculpted to minimize diffraction. There is a single bass port on the back of each cabinet. There is also a pair of standard M6 x 10 mounting holes on each box. My first impression was that the boxes were rather large compared to other monitors with similar driver sizes. I set up the RXA5s in my office and the RXA6s in the studio and spent some time getting to know them both.

LISTENING TEST

My office is a relatively large space with no acoustic treatment. It's where I write, edit video and do graphic design. I usually lis-

ten to music when doing these types of things, so the RXA5s became my entertainment speakers for the few weeks that I had them. I tend to listen to a lot of types of music when I'm working, but some reason, I seem to get the most done when I'm listening to tracks with an electronic rhythm section and synths. Listening to synth-heavy pop, I was quickly impressed by the sound of these monitors.

Like any monitors, the RXA5s got better with age, but right out the box, the upper midrange and top end were really nice. Hats and snares had a crisp quality to them without ever being harsh. The ability to listen for a long time without fatigue is supposedly the draw of ribbon tweeters, but I felt it just as significant to consider that even if I cranked these monitors for short durations, the top end still never felt noticeably bothersome. Vocals cut through the mix really well, and the general separation of sounds in the upper midrange was very clear.

The bass was tight and punchy at first, but as the monitors broke in, they kept their punch but gained some real thunder in the bottom. I had them sitting on Primacoustic Recoil Stabilizers on the desktop, about 8 inches from where the desk intersected with the wall. The interaction of the rear-firing port and this corner seemed to be a pretty healthy recipe for big bottom. Granted, there was that missing low octave that only a subwoofer could bring to life, but for a 5-inch woofer, I was pretty blown away.

All of these factors made for really enjoyable listening, especially when it came to electronically driven pop, hip-hop or EDM. When I checked out more acoustically driven music, I was pleased, but some elements felt a little off. Stringed instruments such as acoustic guitar or piano didn't sound as true to how I know they sound in the recordings. Oddly, despite the ribbon tweeters, there seemed to be a missing airyness. Compared to ADAM A5s, where an acoustic guitar just sounded like an acoustic guitar and a piano sounded like a piano, the RXAs sounded equalized. I was surprised by this, so I swept a tone generator across the top end and found that while the A5s stayed fairly linear up to 20 kHz, the RXAs seemed to have a subtle bump between 2 and 4 kHz, and after that the top end started to gently trail off.

The amplifiers' self-noise was noticeable. When I flipped them on, I would hear it immediately, and while it was easily ignored when referencing styles of music that are consistently loud, the noise reared its head when listening to dynamic classical music or jazz. Also, when listening to jazz, I noticed a bit of a gouge in the lower portion of the midrange and perhaps at a harmonic in the upper midrange. The double bass showed this the most, as the lower body of the instrument was full and present, but the articulation of the plucking and any movement along the fretboard, which was quite clear on the A5s, was more subdued on the RXA5s. I found the same to be true when listening to slightly gritty electric basses in rock music. The low notes were there, but the defining edge was somewhat lost. The body of the snare drum was similarly absent. While the crack cut so nicely and comfortably, the monitors hid the low-midrange frequencies, which suggest the full timbre of the instrument.

BIG BROTHERS

The RXA6s shared many of the characteristics of the RXA5s, with all the comfort and listen-ability in the top end. The studio where I tested them is a smaller space, so the trapping combats the bass buildup relatively success-

fully. I had the monitors on stands about a foot from the wall, and the bottom end did not disappoint. Again, after the break-in period, there was a bump in the low end, probably from the rear port playing off the wall. I started playing with the EQ knobs on the back because I felt like there might be too much bass, but before I got too crazy with that, I decided to bring in a reference for A/B comparisons.

I wanted a fair shootout, so I aimed for a similar price range and put up a pair of KRK Rokit 6s against them. Once I matched levels, it was clear how different the sounds were. I feel like the Rokits aren't perfect, but they are fairly flat and honest throughout the midrange. Their top end is a little hyped and they don't have the bottom end of the RXA6s, but I know them well enough that they make a good touchstone.

I started out listening to some of the things I had studied on the RXA5s and found the same ambiguity when listening to electric and acoustic basses. The Rokit monitors offered more clarity. I tried playing with the EQ settings to fix this, and by turning down the lows and slightly boosting the highs I got a bit closer. By contrast, when listening to hip-hop and EDM, the Rokits seemed muddier than the RXA in the low-midrange, where low-end instruments and vocals overlapped. This compromised the clarity of the vocals. On the RXA6s, vocals sounded much clearer, while the bass synths still filled their role in the bottom end.

As far as imaging goes, the RXAs had an impressive stereo spread, wider than that of the Rokits, though the center of the Rokits was much more defined. Turning down the top end on the RXAs and boosting their gain to match the Rokits helped promote a more defined center, but came at a sacrifice of what air there was in the top end. The center did a good job of staying put even when I moved around the space. Generally, both pairs of RXAs also maintained an impressive amount of detail even when I wandered far off-axis from the sweet spot.

I wanted to see how a mix done on the RXAs would translate to other speakers. I had to touch up some music mixes that had sounded good on my usual monitors, but wound up sounding too bass-heavy when played on a consumer home theater system with a subwoofer. The RXA6s did a good job of highlighting the problem and helping me correct it. I found myself removing some low-lows, and adding a little bit of detail to bass guitar to help it pop. This wound up making the bass sound really good on other speakers, large and small. Overall, I did not make a great deal of changes, but the subtle tweaks led to mixes that translated very well onto other speakers.

THE BOTTOM LINE

When it comes to listening to modern pop music, the RXAs are great. For recreation, and for the price, they are a wise investment. While I do not entirely agree with their "flat frequency response with no hype, just precise audio imaging" assertion, I think that many people would pick them over other monitors because they have rockin' bottom and clarity in the top-end. While they did exhibit some inaccuracies, especially when compared to other monitors, the purpose of a studio monitor should be to give a reference that leads to mixes that translate well to other speaker systems. They RXAs did not fail in that task. The RXA5 and RXA6 monitors are worth a listen. ∎

Brandon Hickey is an audio pro and rabid Blackhawks fan.

Tech // reviews

THREE CONDENSERS TO GROW WITH

Schoeps V4U, Neumann TLM 107, Myrinx Mods

While great ribbon mics are my favorites for unique flavor, they don't always fit into every application—that's where condenser mics shine. For vocals, over a drum kit, as room mics, close up on an acoustic guitar, piano or Leslie cabinet, they expertly cover the gamut. When contemplating a new mic purchase, you always want something you can grow with, meaning it will not only retain its value but its sonic character will stand the test of time.

The three microphones tested here fall into that category. They are all solid players that bring their own taste to the table, are built to last, and in the case of the Myrinx mods, offer something completely new. I powered the three mics tested here with a variety of preamps, from classic Neve 1073s to the Moon 3500MP, a super-clean, audiophile-bred preamp from Simaudio in Montreal.

SCHOEPS V4U

The inspiration for the look of the new small-diaphragm, fixed cardioid Schoeps V4U came from Schoeps' third microphone ever made—the CM51/3, manufactured in 1951. But it's not all about retro chic, as Schoeps states: "The 33mm beveled collar around the diaphragm causes its directivity to increase steadily and smoothly at high frequencies, as in a large-diaphragm microphone." The head can be angled up to 20 degrees front to back, making it easy to fine-tune the placement without turning the stand or mount. SPL is rated at 144 dB, but I never got close to that, nor did the mic ever sound remotely challenged at the output—it was always clear and clean. The capsule is elastically suspended to reduce vibration and the output is a newly developed bridge-type transformerless circuit.

The first application of the mic was on a male vocalist. The V4U is marketed as a vocal mic, but in repeated tests I found that it sounded better elsewhere. Side by side with a vintage U47 as a "B" choice, the V4U sounded great. It was full-bodied and had an even feel to the frequency range on this vocalist, who had a booming voice with a big low end. The main difference between the two mics was at the top. The V4U was a lot more present in the sibilant range, which was overly bright for my taste. I tried to naturally EQ the vocal by turning the mic 90 degrees off-axis, but it was still top-heavy. We recorded a verse and chorus with both mics through the same chain—a Moon 3500MP preamp and an LA2A using moderate to heavy gain reduction at the peaks. After a purposefully blind playback session with the artist, we all decided that the U47 had it over the V4U.

Next, I had the V4U over a Gretsch Gold Sparkle drum kit, powered by a Neve 1073 preamp in a medium-size overdub booth at Blackbird Studio A. It was used as a center overhead flanked by two Telefunken 251s. The V4U gave a great overall picture of the kit, beautifully clear cymbal ring, hyper-real stick hits on snare, toms and cymbals. The bright top of the V4U made the perfect pairing with the 251s. It brought an intimacy to the overall picture that was not there with just the two overheads.

On acoustic guitar, fiddle and mandolin, this mic was in its element. By itself or paired and blended with a Royer 121 on the fiddle and mando, it was, for lack of a better word, "perfect." The same was true when used as a single mic outside a Yamaha C3 grand piano. The attack of the hammers on the strings, even at a distance, was just right, while the rest of the frequency range was equally and realistically represented.

Calling the V4U a "Studio Vocal Microphone" as Schoeps does is misrepresenting its range and unnecessarily underestimating its potential as a great all-around studio stalwart. If you're looking for a vocal mic, look some more, but if you want a great go-to con-

PRODUCT SUMMARY

COMPANY: Schoeps
PRODUCT: V4U
WEBSITE: www.schoeps.de
PRICE: $2,999; $3,069 with Rycote Lyre shock-mount
PROS: Excels across a range of applications.
CONS: Upper frequencies overly present on-axis.

denser that can bring the best when used in a wide range of applications, the V4U is your huckleberry.

NEUMANN TLM 107

The TLM 107 is Neumann's newest large-diaphragm condenser and takes the company's tech up a notch. The compact body and grille design maintains the company's legacy look, and while others have created mics with logic circuits and lights, it's new to Neumann. And it's not just there to bring the bling. When the mic lights up, it ticks one troubleshooting box that is always a question when there's no output at the end of the signal chain: If the light is on, phantom power is home.

Control for the plentiful features is on a slick and tiny joystick at the back of the mic that Neumann calls a navigation switch. Pushing and kicking the stick up, left and right, toggles through the five polar patterns: omni, wide-cardioid, cardioid, hypercardioid and figure-8. You can also switch between no pad, -6dB and -12dB, and adjust the roll-off from none to a sloped attenuation at 40 or 100 Hz. The mic comes with a swivel mount; optional accessories include an EA 4 elastic mount, two phantom-power battery packs or wired power supply.

This mic excelled in all but a few applications. A pair used as far room mics in Blackbird's Studio A brought out the best of a Craviotto Drum Kit when placed 10 feet back and up about 8 feet from the floor. Highly placed in an iso booth above a fiddle, paired with a Royer 121, was beautiful. The Royer brought out the woody body of the instrument without being too "real," while the TLM 107 brought out the detail of the instrument just enough when tucked in under the Royer in the mix.

On acoustic guitar the pair of TLM 107s put the player right in the room with the speakers. It was very easy to go in and switch the patterns, and it was great to have the confidence of knowing that the mic was getting P48 when the lights were on.

Other applications included the pair over a drum kit, on a male vocal and on a bass cabinet, with only the latter being a poor choice. In this case, the amp was an Ampeg B15, and while the TLM 107 sounded okay, it just didn't bring the best of that amp to the session. For most sessions we've used an AKG D112 or an EV RE-20 in this application, and that is where we ended up here.

We had the opportunity to bench test the TLM 107 with our Stanford Research SR-1 audio analyzer (available for download at mixonline.com). The tests reflected Neumann's own published tests.

In my work I use many Neumann mics, new and old, and the TLM

107 represents the best of what Neumann produces—sonic parity with their legacy mics, plus innovation. The navigation switch is an excellent way to switch patterns, roll-off and pad, plus it gives you phantom-power confidence. Add to that the fair price, and you have a mic that can't be ignored when looking for an investment you can grow with.

MYRINX MODS

Myrinx is a brand new company out of Switzerland that offers a different take on mic design and implementation. (The company also makes high-end cables under the VOVOX name.) Myrinx's "product" is to modify existing microphones. The company's current, and expanding, product line comprises five models: the Brauner Phantom and VM1 tube mic; Microtech Gefell M930; Neumann U87; and Pearl CC22.

What Myrinx promises to do is improve the microphone beyond its manufacturer specs by completely renovating each unit. You provide the mic, Myrinx does the mod. Myrinx's Jürg Vogt described the approach to me: "Imagine you could listen to a capsule without a housing, like it would float in space, and listen to the signal coming from the capsule directly after the PCB—with no cables in between. That's the ideal we try to approximate."

To accomplish this, Myrinx replaces all internal wiring with purpose-built, solid-core, ultra-pure copper wires coated with a mesh of natural fibers (no plastic coating) from VOVOX. Next, they remove all unnecessary connectors, soldering the connecting cable directly to the PCB, and then separate the shielding and audio ground as much as possible. The metal housing is replaced by European walnut internally shielded with copper foil. Wood has been chosen because it is non-conductive, won't hold an electrostatic charge, is a poor resonator, and has little interaction with the electric signal in the microphone. The unique, and replaceable, grille has no parallel surfaces and is made from organic lignin fibers, which is infinitely irregular, has open cells for excellent transmission of audio waves, and is rigid, non-resonant, and durable. Beneath the grille is a high-precision metal grid designed and made in Switzerland.

For this review, I had a copy of a Brauner Phantom that had been Myrinx-ised. It came in an attractive wooden box with a space for the mic, a Rycote universal mount, and a wind-around channel to hold the integrated cable. The mic is incredibly light and attractive. The lignin fiber top slides off with some effort, meaning it's not easy to lose and can be user-replaced after some wear. To me this is a bonus when you're

intelligibility is enhanced. But I've never "heard" a great bench test, and as all things in art, life, and audio, beauty is in the eye of the beholder, so you should hear it for yourself. I loved the tracks recorded with the Myrinx Phantom mics. They were the epitome of high-end recording. When I use the Moon 3500MP in a signal chain, I know from experience that what I'm hearing is not being colored by the preamp—in this case the mic and the preamp were doing the same "hands-off" dance to my ears. Myrinx has a new idea and it's worth a listen. ■

using the product around a variety of singers. Not only is it cleaner, it makes you wonder what gets into other mics where you can't remove and replace the grille.

I'm a big Brauner mic fan. They make products that sound unique and are great go-to mics around the studio. The pair of Myrinx Phantoms I tested were no exception. On lead vocals, acoustic guitar, over a drum kit, and on acoustic piano they were excellent and vacant at the same time. Hyper-reality to me means that a mic and signal chain is not adding color to the listening experience. The Myrinx Phantoms, powered by the Moon 3500MP dual channel preamp, provided this. I used the Phantoms in combination on a male lead vocal and acoustic guitar on the same project. The tracks were clean, clear, and gave me back just what was happening at the performer. While this approach doesn't work for all types of music, it's something you want to have in your bag of tricks when that situation arises. The pair over a drum kit, or placed back from a kit at knee height about 9 feet, offered a great rendition of the drums close and far—beautiful.

As an extra case study, I had two Brauner VM-1s, one pre and one post Myrinx. You can see the comparison tests we ran on these two using our Stanford Research SR-1 audio analyzer. Across most patterns, the Myrinx VM-1 exhibited 1-2dB hotter output in the 100Hz and 4-5kHz range. In some cases, this became less evident off axis. Download the reports to see the analysis firsthand at 0, 90, and 180 degrees.

So does the Myrinx concept translate to the end product? Is it better? The bench tests of the VM-1s show that there is improvement in two critical ranges where you could argue that warmth and

NATIVE INSTRUMENTS RISE & HIT

Explosive Cinematic Sound Library and Virtual Instrument

Fig. 1: Rise & Hit offers controls for riser synchronization and latch mode, a waveform view and more.

very fan of thrillers, action films, and horror and sci-fi flicks has experienced the emotional impact of a deftly executed riser and hit in a cinematic score. But all you have to do is listen to the grand finale for the Beatles' "A Day In The Life" to realize how electrifying the technique can be for popular music, too.

Designing fresh sounds for and synchronizing a rousing riser and hit can be a time-consuming process. Thankfully, Native Instruments Rise & Hit makes it easy, inspiring and fun. The included, deeply layered cinematic sample library makes dramatic and complex sound design drag-and-drop child's play. And the integral virtual instrument makes synchronization a breeze: It lets you specify exactly how many seconds or tempo-synched beats the riser will take to go from start (MIDI Note On event) to apex, regardless of the native length of the samples you use.

GETTING IN KONTAKT

Rise & Hit is a Kontakt Instrument compatible with NI's Kontakt 5 and the free Kontakt Player. The 8.6GB library (compressed to 6.25 GB using KONTAKT's lossless compression) contains 4,250 samples, mostly recorded at 24-bit/96kHz resolution and down-sampled to 48 kHz. Two samples—one for the riser (or rise, in NI-speak), the other for the hit—constitute a layer, and a single NKI instrument can comprise up to four layers. MIDI notes C-2 through C#3 trigger the rise portion of an instrument, and D3 through G8 trigger the hit. (The two ranges are colored differently in Kontakt's virtual keyboard to reflect this split.) Included are 750 layer presets (80 with effects processing) and more than 200 preset NKI files (instruments).

The main page of Rise & Hit's GUI is where you set up rise synchronization for the currently loaded instrument as a whole; select in turn up to four layers for editing their rise and hit samples' relative volumes, rise start-time offsets and hit decay times; and view the currently selected layer's dual-sample waveform (for the combination rise and hit; see Fig. 1). You can sync the rise time (that is, its duration) for an instrument to the host's tempo and set it to last the number of beat subdivisions (for example, quarter-notes in 4/4 time) that you specify. Alternatively, you can set the rise time in seconds, irrespective of the host's tempo. Activating latch mode plays the entire sample, even if you don't sustain a MIDI note for the duration; in fact, I found Rise & Hit played to the end of the sample in latch mode even when I stopped Digital Performer 8.06's transport immediately after a pre-recorded MIDI note began. Activating a Continue button automatically makes the hit (for all layers) voice at the end of the rise samples when Rise & Hit receives a MIDI note in its lower (rise-

TRY THIS

To create an especially dramatic riser in Rise & Hit, load four layers for an instrument and stagger their offset and volume controls so that each successively voicing rise sample is louder than the preceding one. The lower-left portion of Fig. 1 illustrates this setup.

assigned) register.

The quickest way to combine sounds is to select factory-preset layers from the Layer Browser, as these each load a complementary rise and hit sample. Clicking on a magnifying-glass icon in any of the four layer slots on the GUI's main page provides access to the Layer Browser, which is arranged in categories including orchestra sections, voices, bells, electronics and more. Click on a tag button above the browser to further refine the displayed presets to include only those layers that, for example, sound natural (of acoustic origin) or processed or execute a glide (typically upward) in pitch. You can also load an empty layer for adding your own combination of factory samples; more on this in a bit.

Back in the main page, the individual layers you've loaded can each be soloed, muted or deleted. By dragging to the right an Offset slider for a layer, you delay the start of its rise sample with respect to the other layers' start times and compress its length accordingly. Clicking on a layer's Whoosh button purportedly loads a shorter, complementary sample that serves as an enhanced attack for the ensuing hit; to my ear, it simply shortens the envelope of the currently loaded rise sample and offsets it so that it doesn't voice until the last moment before the hit sample voices. The layer's Decay slider adjusts the length of the hit. Two other sliders adjust the respective volumes of the layer's rise and hit; they can be linked.

DIGGING DEEPER

You can maximize the view of a layer to reveal additional controls that adjust the panning and tuning of the layer and add delay, convolution reverb, EQ, parallel compression, limiting and distortion via effect sends (see Fig. 2). (Parameters for EQ and the dynamics and effects processors are adjusted on the Master FX page, which you navigate to from a tab at the bottom of the GUI.) Raising a crossfade control purportedly causes the transition between rise and hit to be progressively blurred, but I found its effect to be so subtle I wasn't always certain it was working.

You can also modulate any mix or effect control for a layer. For example, you can create an S-shaped curve for the rise's volume that peaks at the transition point into the hit. If the effect sounds too overblown, lower the modulation intensity. Modulation for panning and tuning is bipolar, changing value both above and below the current slider position.

As I alluded to earlier, you can create your own custom layers by combining any two factory-supplied samples (one rise and one hit). The Sample Browser—available from the waveform section—is organized into categories in much the same way as the Layer Browser. The Sample

Fig. 2: Expanded view has controls for panning tuning, modulation and effect sends.

Browser also provides a button that automatically loads a complementary rise and hit at once when one or the other sample is selected.

RISE & SHINE

Rise & Hit's library includes a wealth of excellent sample recordings of orchestras, string and percussion sections, choirs (mostly vowel sounds), bells, cymbals (including reverse-envelope), piano, guitar feedback, electronic sounds and various effects (Doppler, reverb and noise)—many with heavy processing. Most of the content is oriented toward cinematic applications, but I can imagine it also being put to occasional use in, for example, EDM.

When you load a layer preset while synched to host, the layer's volume, offset and decay controls—as well as the instrument's rise time—remain locked to the values you set previously. That's a good thing because it lets you audition different sounds while maintaining the envelope you deliberately devised to lock to picture or a song transition. I also liked that I could adjust decay times lower for select layers to morph the relative balance of the instrument's composite sound toward its end. To trigger only hits, without a preceding rise, I pressed a D3 note or higher on my Roland A-37 MIDI keyboard controller.

On the downside, I wished latch mode was a global setting for instruments; every time I loaded a new instrument, latch mode would turn off, making auditioning successive instruments using my keyboard (as opposed to using a pre-recorded, sustained MIDI note) more tedious. (Fortunately, latch mode stays locked when loading successive layer presets.) Linking a layer's volume faders nullified any existing offset between the two, defenestrating my carefully wrought balance of rise and hit. I was also disappointed that playing different MIDI notes on my Roland A-37 didn't transpose the pitch of Rise & Hit instruments on playback. Of course, you can always tune an instrument in Rise & Hit's instrument header, radically so if you wish—plus or minus three octaves.
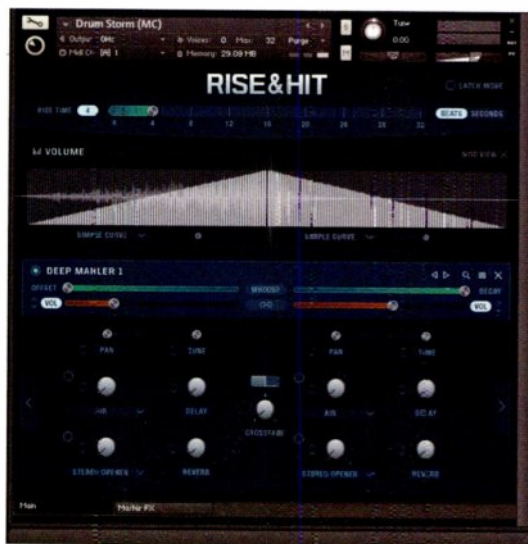
A BIG HIT?

Rise & Hit's main application is cinematic sound design, for which it is a great creative aid and huge timesaver. The sample content is wide-ranging and sounds great. Although the GUI can use a little refinement, it is mostly very well designed. Thumbs up! ∎

Mix contributing editor Michael Cooper is a recording, mix, mastering and post-production engineer and the owner of Michael Cooper Recording in Sisters, Ore.

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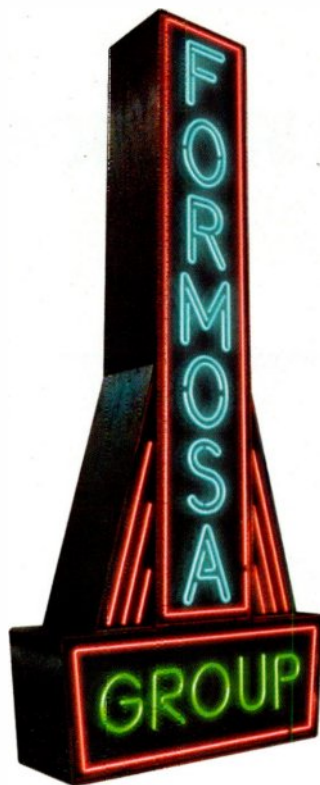
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Tech // reviews

THERMIONIC CULTURE PHOENIX HG15

Classic Valve Compressor

Thermionic Culture is known for shaping sounds to extremes, and the new single-channel HG15 is no exception: it goes to 15!

Thermionic Culture is known for producing high-quality pro audio pieces that offer atypical sonic qualities. My first experience in using its gear amazed me at how much I could alter the sound of the source with one device. When I learned the company was building a single-channel unit based on the company's classic Phoenix stereo compressor, I was interested in running it through its paces. The Phoenix HG15 valve compressor comes with a twist—it includes a preamp with phantom power, variable control settings based off of a Vari-MU design, and limited EQ settings and bass cut to mimic the idea of almost being a full-on channel strip.

SIMPLE DESIGN

The HG15 is a striking two-rackspace, rust-red unit with eight black knobs, three metal switches and a large vintage-style VU meter. Features include input and output detented knobs, a phantom power switch, bass cut and a sidechain to reduce proximity, air and presence switches, and six-position time-constant control designating how much compression is applied. Although I knew it was a tube unit (two valves: 6AQ8EH input and 12AT7/ECC81 output), it wasn't as heavy as I expected. The build is solid and simplistic; the knobs are made of plastic but feel sturdy to the touch, and I always appreciate the detented knobs for input and output control, which makes for easy recall. The small metal

switches are solid but took a minute to get used to, as I had to push them down where most other units are pushed up to engage.

The back panel is even simpler, with single XLRs for input and output, IEC connection for 115/ 230-volt power, and a single phono connection for linking another HG15 for stereo use. The HG15 does not get too hot when in use, so stacking the unit in a rack between other units is not a problem. The input gain/impedance is variable when switching between sources—Mic: 52 dB/1kΩ; Pad: 38dB/2KΩ; Line: 25dB/10k; and the Output Impedance is fixed at 600Ω.

JUST A COMPRESSOR?

Getting in and out of the HG15 is simple by using the XLR connections provided on the back panel. There are three choices on the front end of the HG15: Mic, Pad and Line. It plays well with most tube and condenser mics, with the additional phantom power supplied. The HG15 maxes out at 52 dB of gain, so a Cloud Lifter or something similar would be helpful to get ribbons or low-impedance mics up to a level that isn't noisy. I found success

TRY THIS

Use an XLR-to-¼-inch turnaround cable to connect an instrument cable to the HG15. Plug in your favorite bass and drive the input enough to achieve some valve distortion. Apply compression to taste and experiment with the EQ/bass cut switches to allow the low-end to rumble along nicely. Who needs a distortion pedal when you have an HG15?

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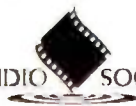
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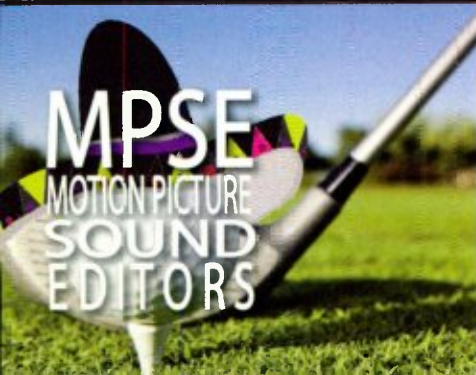
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using the Pad position with a couple of my tube mics on louder input sources, such as a mono drum mic or electric guitar, whereas the -14dB pad helped the input stage stay smooth and not peak the front end of the compressor. When using the HG15 as a stand-alone compressor either in tracking or mixing, the Line position was the best choice, operating like the original Phoenix compressor but with slightly harder compression.

The compressor design in the HG15 is similar to a Vari-MU or Fairchild 670 with fixed threshold positions, one being minimum, five being maximum. The ratio is not user-definable, as the amount of compression increases toward limiting stages when the input levels increase. The six positions of time-constant control allows for flexibility in the HG15—fast, medium and slow, with variations of that in positions 4, 5 and 6 (1 = F/M, 2 = F/S, 3 = M/ F, 4 = M/M, 5 = M/S, 6 = S/M). Pushing vocals aggressively through the HG15 yielded wonderful results. Additional tone shaping can be applied with variations from the presence and air switches, which will lift midrange and high levels of EQ between 3 kHz and up to 18 kHz. Both can be switched in and out independently or by simply turning on or off the Active EQ switch. The bass cut can be applied to allow signal of the frequency selected to pass without the HG15 applying as much compression, which is helpful when close-miking an acoustic guitar, vocal or guitar cabinet to lessen proximity from the direct source.

PHOENIX IN FLIGHT

I spent a bit of time getting to know the front panel of the HG15. I was most curious about the sound of the preamp, so I tried a few different input sources for comparisons. The preamp in the mic position is clean and open sounding. It can also be driven hard to emulate the '60s and '70s drum sounds that are sampled so heavily today. The HG15 quickly became my "channel strip" of choice for this situation; the combination of the preamp and the compressor added the perfect amount of color to either of the phantom-powered Royer 122 or Fathead II, as well as a Mojave 301 and 201 FET condenser. My Peluso 251 tube required using the pad when recording electric guitar, yet still sounded open and full. Tweaking the amp's settings just a bit allowed me to not have to apply any EQ. Recording an acoustic guitar with the same mic, I added the EQ switches as I wanted a bit more air for some shimmer above 10k, which worked well.

I enjoyed running bass signals through the HG15, either guitar or a Moog Little Phatty keyboard. I had to use a ¼-inch-to-XLR turnaround to get into the unit, but as long as you have one handy, the HG15 handles bass surprisingly well. Driving the preamp created wonderful fuzz bass tones, and I used the compressor output to control the signal to the DAW.

No matter how aggressive I was with the pre or the compressor, it still sounded useful. The preamp also shined on acoustic guitar and female vocals, and adding the subtle EQ switches and/or bass cut, I was able to find the sound I was looking for. It would be great to have

an additional switch that allowed the user to define where the EQ fell in the signal chain, a feature that I enjoyed having on my Pendulum Quartet II channel strip.

On vocals, I could get a wide variety of sonic detail depending on how hard I drove the input side—the two different tubes complement each other nicely. I tend to lean toward medium attack and faster release times when tracking vocals, then a little slower when using a unit as a second compressor to even out the recorded signal. I used it on a female lead vocal for Nashville-based Yumi and the System. The singer wanted her vocal to be slightly gritty, but not sound like a full-on distortion pedal. In each track I was able to vary the tone slightly to help the vocal have a signature sound without being too harsh or over-compressed. Having the preamp and compressor in one unit allowed me to find an even balance of saturation, and the EQ switches helped the vocal cut through the mix with the right amount of presence.

I found myself wanting to switch the meter between input and output to see how hard I was hitting the HG15, but the meter is not selectable, and shows only how much gain reduction is being applied. I learned to listen to the unit and back it off when it was feeling too aggressive for the song. Applying the HG15 on bass when mixing sounded fantastic, as well. The slower release times helped control the bass in the track without sacrificing low end. One of my favorite applications when mixing with the HG15 was to create a mono aux send and send different levels of drum inputs to the aux channel. I would then drive the front end of the HG15 and create some dirt to mix in underneath the dry close mics.

The Phoenix HG15 from Thermionic Culture fits right in line with the company's quest for creating products that provide different colors to add to your workflow.

USEFUL CHANNEL STRIP

The Phoenix HG15 from Thermionic Culture fits right in line with the company's quest for creating products that provide different colors to add to your workflow. It did not describe this unit to me so much as a traditional "channel strip," but I found myself using as such in many situations. The preamp is of high quality and sounds great on its own. The compressor, based on the company's original Phoenix stereo compressor, is easy to use, and with the Vari-MU/Fairchild 670 design, the user can apply it with varied degrees of compression up to hard limiting.

Its EQ is simple but very useful when searching for ways to help an individual track sit well in a mix. The only feature I didn't get to try was linking two HG15s, which could prove to be a powerful stereo pair. If you're looking for a piece of gear that offers more than you'd expect, the HG15 provides plenty of musical-sounding options for tracking and mixing. ∎

Chris Grainger is a producer/mixer/engineer and owner of Undertow Studio in Nashville.

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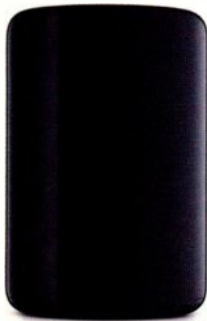
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My Little Helpers

By Kevin Becka

I remember the first time I heard the term "mobile apps." It was 2005, in the pre-smartphone era when *Mix* editor Tom Kenny asked me to hunt down some candidates for a feature. After much searching, and trying to imagine the potential for portable audio companion apps that in large part hadn't yet been defined, I found there wasn't much that was useful or mobile. Now, I don't go a day without using one or more as a helpmate for engineering, writing, teaching or just living life.

The app I'm most excited about these past few weeks is the new ARATechLabs Arapolarmic app for the iPad. It lets you use an iPad's camera and display to monitor the directivity of microphones in real time. You just match the "real" mic's orientation with a picture of it from the gallery, and once the app syncs, it displays a movable and scalable overlay of the polar pattern.

I'd been following the progress of the app, which at first was just YouTube videos, and was texting back and forth with Michel Guerra from ARATechLabs. The company's name is part acronym, which stands for "Augmented Reality Audio Technology." At first, the name set off my BS alarm until I got my hands on the app. It's not so much for pros who understand polar patterns and how they shape the tracks we record. But for teaching, it is unparalleled in how it makes something largely imagined and invisible, tangible and instantly understood. Just last week I used it with a group of high school students in for a summer recording camp at The Blackbird Academy. You could see the light bulbs popping on as I moved the iPad around a Royer R-121 and explained how a figure eight pattern captures the sound around the mic. "No Way!" Way.

Arapolarmic is not just the Philip K. Dick stuff, though; there is a page with full specs on each mic, output voltage graphs and polar diagrams across a number of frequencies, a picture gallery and more. As with early versions of anything, it's sometimes clunky and the list of mics available on the app are limited. But having AEA, Audio-Technica, Audix, Royer, Beyerdynamic, EV, Lewitt, Mojave, Schoeps, Telefunken and Violet onboard for starters is pretty darn good. Teachers, keep your eyes on this one.

> *Now, I don't go a day without using one or more [apps] as a helpmate for engineering, writing, teaching or just having a life.*

Last month I wrote about my switch from an iPhone to the new Galaxy S5, which means I had to find Android equivalents of all my favorite Apple-compatible audio apps. The good news for Android users is that there are some great, and mostly free, apps to be had. For example, when you're tuning vocals and need the key of a song, the simple octave-and-a-third Perfect Piano is, well, "perfect." It's designed for giving piano lessons, but for key-hunting it works great, and even has the names of the keys and octaves if you're not a player.

Echolalia is free and beats iPhone's BPM hands down. It lets you tap or type in a tempo and gives you eight delay breakdowns in milliseconds, from a whole note to ¹⁄₁₂₈th note. The GUI is simple and fast, there are no ads, and you can easily change the delays to triplets or dotted notes.

RTA Analyzer from RadonSoft is free and usable as is, but RTA Pro (3.99 Euro) offers 40, 60 and 120 bands, permanent calibration, different peak-hold times, adjustable integration time, RMS/peak/freq display options, and you can freeze and save the image. Of course you're limited by the phone's microphone but who knows where this will go? RadonSoft makes other useful apps including a spectral audio analyzer, a signal generator, and a noise and sweep generator.

ProAudio Tools is a decibel meter calibrated with a B&K Type 2236 Sound Level Meter. It has a flip screen option for better audio capture, selectable time responses, calibration function, and also offers the ability to show/hide common SPLs. From the odds 'n' ends department, PitchLab and DaTuner Lite are two tuners that can come in handy when the talent doesn't come prepared.

Not to say that Apple doesn't have some fantastic options also. Audio Tool from Performance is a bundle of six apps that offers a decibel meter, tempo calculator, tone generator, sample/bit, drive space calculator, and an app that turns your iOS device into an ad hoc talkback mic with an on/off switch and more. It's $1.99

Having all these great tools in your hands while you teach, record, mix, overdub and communicate with musicians is a great way to offer more to the client, communicate with the next generation of engineers, and bring more tech to your sessions. And when you're done, you can call a friend to tell them about your new portable audio toys. ■

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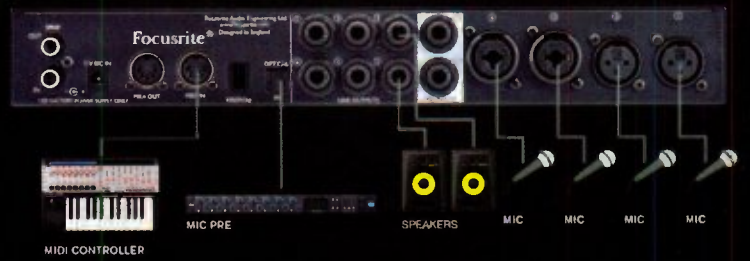| FOUR Focusrite Mic Preamps | FireWire & Thunderbolt Compatible* | Expandable with ADAT & S/PDIF (18 in/8 out) | Pristine 24-bit/96kHz Conversion | Two Independent Headphone Outputs |

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