# 2018 NAMM Show Report
## Mike Rivers © 2018

***Anaheim, California, January 25-28 2018***

It's that time again, the largest musical toy show in the US. If I haven't miscounted, this is my 30th Winter NAMM show. Along with changes in trends in the music industry, there have been changes in the focus and structure of the show over the years, but this year's show has had some really significant changes from recent years. If you're reading this report, it's probably because you're interested in new products and developments related to recording and sound reinforcement, and that's where I've seen the biggest changes in how NAMM presents them, and they're all good.

The Anaheim Convention Center has had a new section under construction for the past three years. It's now complete, so naturally NAMM had to fill it. While a few long time exhibitors of audio related products held on to the show floor spaces that they've occupied for many years, nearly all of the audio exhibitors moved, either voluntarily or with a little coercion, to the two floors in the new North Hall, resulting in what felt like a slightly scaled down version of the exhibits at an AES (Audio Engineering Society) convention. It was so much easier to carry on a civilized discussion at a booth, or listen to a microphone or equalizer, without having to compete with five people nearby banging on drums or playing bass riffs. There were a few in-booth performances in this section, but they were pretty tame and tended to be fairly short (there's probably a policy to support that). The North Hall is a pretty long hike from the main halls, but I never got the "Gotta get out outa here and rest my ears" feeling.

NAMM has always offered a fairly robust educational program, and this year (and at least for next year, I believe), they've formed a partnership with AES to augment their usual business-oriented training sessions with a program of technical classes on topics relative to pro audio. I didn't attend any of these classes myself, but the AES folks I spoke with seemed to be pretty happy with how it was going, particularly for a first attempt.

Another big change with the show this year was enhanced security. In the past, security guards checked badges and sometimes a picture ID at the entrances to each hall, but this year they moved security to the lobby entrances, set up magnetometers and bag checks (no x-rays yet) and the lines to enter the show became very long very quickly, particularly on the first two days. It's unfortunate that this indeed appears to be a legitimate concern, but hopefully security procedures will go more smoothly in the future.

There aren't many game-changers this year. It's getting harder to tell the new and innovative from the "me too." But anyway, on with the show.

*Microphones, Preamps, and Mic Accessories*

The usual suspects were here, but this was a bit of a lackluster show for new products, perhaps because many showed their latest wares just a few months ago at AES, or at last year's NAMM show, with development moving a little slower than anticipated. What's interesting is that I've noticed some bragging about transformer design and name brand transformers (Lundahl and Sowter, for example) incorporated into new mics. It's an interesting topic and it's good that they're paying attention and bringing what they've learned to our attention.

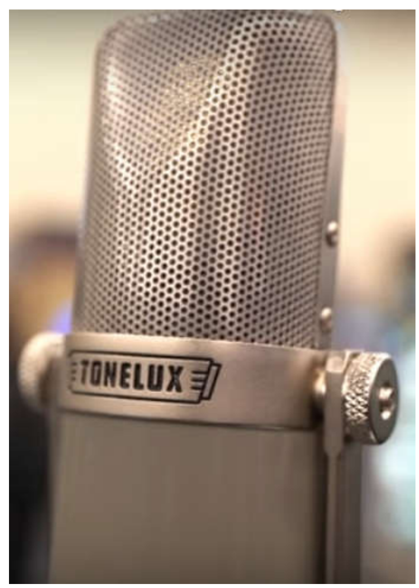
The big buzz in mics this show is that Neumann is re-issuing the legendary U67 large diaphragm three-pattern tube microphone. Originally introduced in 1960, with production continuing through 1971, the 2018 version is built and tested to the original specification, using original materials and parts including the original transformer and EF86 tube. They're confident that a 2018 U67 will perform just like a new one from the original production. While I've certainly heard recordings made with a U67 (whether I knew it or not), I've never had the occasion to use one myself, and probably don't have a compelling need, but there's no question that this microphone has a sound and well-deserved reputation.

The 2018 U67, like the original, comes as a kit with a padded case, power supply, an elastic suspension mount, and cables. While the mic is completely true to the original design, the power supply design is new, compliant with today's electrical safety requirements, operable from worldwide line voltage standards, and is able "to accommodate the slightly higher filament current of newer premium-grade tubes." This suggests that users may want to experiment with tubes from different manufacturers, or, that Neumann has selected a modern tube with a slightly higher filament current requirement than the original, or both. The new power supply is fully compatible with older U67s should you have an old mic with a ratty power supply. Although not published, I suspect that they'll sell a power supply separately.

All microphones, like people, change with age. If Neumann has done their job, there's no doubt that those who recognize the U67 sound will find the 2018 model to be in the same family, though there will always be some who will say that it doesn't sound like the one they own or rented for their last project. Also, a young engineer recording with U67s today likely only knows the sound of a mic that's aged more than 40 years, so a virgin re-issue may or may not have the sound with which he's familiar. But with all of the new mics available that are patterned after the U67, I suspect that the new Neumann will come closest to providing what those who want that sound are after.

Bringing a new batch of U67s into the market, even at a fairly stiff price, will make them more accessible to those who want the sound. While there will be collectors who want nothing but the original issue mics, the used U67 market, which, like with many vintage audio products, has gone sky high, might stabilize a bit.
http://www.neumann.com/?lang=en&id=current_microphones&cid=u67-set_description

Another re-creation of a slightly older vintage mic, the Sony C37a, appeared at the PMI Audio booth under the Tonelux badge. Originally introduced in the US in 1958, as Sony's first tube condenser mic, it was popular in studios in the 1960s, particularly for big band recordings and big band vocalists. The origin of this project was a discussion between engineer Joe Chiccarelli and Brent Casey, the man behind PMI's Studio Projects series of condenser mics. Joe, who incorporates a lot of the 1960s-era recording techniques into his projects, asked if it would be possible to build a mic today that had the characteristics of the C37a, one of his favorite mics.




Brent took up the challenge and over about a 5 year period of experimentation with capsules, impedance converter circuits, and especially transformers, came up with a mic that Joe was happy with. The transformer was probably the biggest part of the project since originals were no longer available, nor was the alloy used to make the original transformer laminations. Brent understood that there's more to a transformer than the proper turns ratio. DC resistance, inductance, and capacitance contribute to a shaped frequency response. The transformer's core material, lamination design, and orientation are what determines how much and what kind of non-linearity the transformer has – and that's where the color and dynamic response come from.

Two mics came out of this research and development project. The JC37 is a side address cardioid tube mic with a contemporary design power supply that can power two mics should you want a pair. Note that the original Sony had a switch on the back to change it to an omni pattern, but since nearly everybody used it as

a cardioid, Brent decided to eliminate the switch. The VC37 shares the same components except for the impedance converter, which uses a FET instead of a tube, and is phantom powered. The capsule is custom built, using a diaphragm mounting scheme that Sony used in their capsule, which is somewhat different than today's capsules. Expected delivery is before Summer 2018, at the projected, and very reasonable prices of $1500 and $2500 for the FET and tube versions respectively.
http://tonelux.com/ No info here yet, try again later.


The Royer R-10 was introduced around the time of the 2017 Fall AES show, but since they weren't there to show it, this is its first appearance at one of the shows I attend. This is Royer's least expensive ribbon mic, just under $500, and all reports are that it's a great performer. It uses the same ribbon and magnet assembly as their R-101, with Royer's signature "offset" ribbon, giving a slightly different tonal characteristic front and back. The benefit is sort of like getting two mics in one, with the downside being that it's not the best mic to use as the side mic in an M-S setup.

The primary differences between the R-10 and R-101 that result in the substantial saving are in the metalwork and the transformer. The body of the R-10 is lighter than that of the R-101, the grill design is different, and it has an attached

swivel mount rather than a clip or elastic suspension. Since I had read in an R-10 review that the transformer was the greatest contributor to the cost saving,

I had a chat at the show with David Royer, who designed the R-10 transformer, about what makes an expensive transformer expensive. Turns out that a significant part of the cost is in how the core is constructed, which affects how the windings are applied. There's considerably more hand work involved in winding the R-121 transformer than in Royer's R-10 design. The R-10 transformer has a lower turns ratio, resulting in about 6 dB lower sensitivity, which in turn allows the mic to take a higher SPL before the transformer begins to saturate unpleasantly. Ribbon mics seem to have found a place in the studio and stage close to the front of guitar amplifiers and over a drum kit, both high SPL applications, and according to reviews, still gives good performance on acoustic instrument when used with a quiet preamp with plenty of gain.
http://royerlabs.com/r-10/

I think the Integral close miking system from Samsystems in the UK was a last-year product, but it was new to me and I thought it was kind of clever. It looks a bit like a go-cart steering wheel with a cardioid dynamic mic element mounted in the center. The outer ring is sized and drilled to match the mounting pattern of a 10- or 12-inch guitar amplifier speaker. It's installed inside the cabinet between the front panel and the speaker, which positions the mic a couple of inches in front of the center of the cone. A cable with an XLR connector is provided to connect the mic to a mixer or preamp. This gives you a consistent mic and placement for quick setup and predictable sound to work with. Dead-on a couple of inches from the cone isn't my favorite miking position, but I suppose that in the interest of time and consistency, whether you're on tour or setting up in the local club, you can make it work. In the studio, you can always supplement it with additional distant or room mics.

https://www.samsystems-uk.com/

It's hard to keep track of what's new in mic preamps but one that got my attention was the SFP-60 dual channel tube mic preamp from a new (to me, anyway) company called Useful Arts. Its shtick is a "Color" control that adds second harmonic distortion with the goal of making vocals sound clear, forward, and intimate. It's interesting to note that the Aphex Aural Exciter, introduced in the mid 1970s and for a few years was used on just about every pop recording of the period, employed 2nd harmonic distortion to do its thing, which was primarily to improve the clarity of lead vocals.


Another design feature of the SFP-60 is that it uses what they call a "choke loaded" output stage. A traditional single-ended tube output stage has the primary of the output transformer in series between the power supply and plate of the output tube, with the output taken from the transformer secondary. The choke loaded design puts a large inductor (the choke) in series with the plate rather than the transformer primary. The output transformer is coupled to the plate through a capacitor. What this buys you is that the transformer has no DC passing through it, and any saturation in the transformer results directly from the audio and is independent of the fixed DC current. Both the choke and the transformer contribute in different ways to saturation distortion and having two designable parameters offers more flexibility in getting a desired sound. This configuration is used in the Telefunken V76 mic preamp, and is likely a contributor to its desirable coloration. Front panel switches, which control relays rather than passing audio through their contacts, engage a low cut filter, a pad, invert polarity, switch 48v phantom power, and select between mic and instrument DI input.

The same basic circuit is used in Useful Arts' BF-1 tube instrument DI, a single channel unit with a fixed amount of "color," a two band equalizer, and an input impedance of around 20 megohms.
https://www.usefulartsaudio.com

In the area of microphone accessories, Triad-Orbit, who has been making some innovations in mic stand design and construction, has licensed the Starbird boom stand from its previous builder, Manley Labs. This is a very large, heavy duty stand that's ideal for getting heavy mics up high and keeping them there. A few significant improvements over the original Starbird design (which, by the way, is named for its designer, George Starbird) include legs that fold inward to reduce space required for storage and make carrying easier, a positive boom angle lock using a spring-loaded pin, and a pneumatic booster to assist in raising the boom and lowering it gently. There's also a clever counterweight assembly that has a hollow chamber that can be filled with lead shot for additional weight when needed to balance a heavy mic on a long boom extension. Beefed up versions of the Triad-Orbit patented clamps, clutches, swivels and quick-change mounts are included in the updated design. With a maximum vertical height of 102 inches and boom length of 98 inches, it's possible to get a mic more than 16 feet up in the air.

Also new this show from Triad-Orbit are a couple of microphone clamps, an adjustable screw clamp good for attaching a mic to tubular structures such as a lighting truss or the tube of a mic stand, and a spring clamp similar to a welding clamp for clamping to flat surfaces.
http://www.triad-orbit.com/starbird

***Computer Audio Interfaces***

I was enthusiastic when Focusrite brought out the Clarett line of interfaces a couple of years ago – functionally very similar to their very good Scarlett line but with upgraded mic preamps based on their ISA series, and converters derived from their high end Red series. However, their computer connection was through Thunderbolt, which left me, a PC user, un-Claretted. I've been pestering Focusrite ever since to bring out a USB-connected Clarett, and at this year, they finally did it. The three Clarett USB models line up pretty closely to the Scarletts, with the number of mic preamps corresponding to the model numbers, the Clarett 2Pre USB, 4Pre, and 8Pre, all of which are capable of recording 24-bit audio at up to 192 kHz sample rate.


The 2Pre and 4Pre are desktop boxes while the 8Pre is a 1-space rack mount unit. The smallest of the three, the 2Pre, is equipped with two front panel mic/line inputs on XLR-1/4" combo jacks with inputs 1 and 2 switchable to high impedance instrument DI inputs. There's a headphone jack with its own volume control, and a monitor volume control that can be assigned through a software control panel to any or all of the outputs. The rear panel has four line outputs, MIDI In and Out, and an ADAT optical input for up to 8 additional inputs. Power is supplied by an external power supply.


The 4Pre and 8Pre offer, respectively, 4 and 8 mic/line inputs, with inputs 3 through 8 being on the rear panel. Both have two front panel headphone outputs, S/PDIF coax in and out, and. The 4Pre has four additional rear panel line inputs and four line outputs on the rear panel. The 8Pre has 10 analog outputs plus an ADAT optical and word clock outputs. Each of the combo jacks has its own gain control, while the line-only inputs are at fixed gain. Metering on the desktop models is with Focusrite's multi-color ring of LEDs surrounding each of the input level controls. The 8Pre has 6-step LED ladder meters for the eight analog inputs and main stereo output.

USB 2.0 or above is required, and both USB-A and USB-C cables to accommodate both PCs and Macs are supplied. Bundled software includes the Focusrite control panel for setup, routing, and input monitor mixing, a few plug-ins, and, as of this writing, the Abelton Lite DAW software.
[https://us.focusrite.com/news/clarett-usb](https://us.focusrite.com/news/clarett-usb)


Universal Audio introduced the Arrow, a scaled-down version of their Apollo interface line. Housed in a desktop box, it has a similar but simplified control panel, two inputs and stereo main and headphone outputs. It inherits the Apollo converters and preamps, including UAD's Unison mic preamp modeling. A UAD-2 Solo hardware processor supports Universal's plug-ins for DAWs, a few of which included with the Arrow are their Teletronix LA-2A compressor, 610-B preamp and EQ, Pultec EQP-1A, and a collection of guitar and bass amplifier models from Softube. The computer connection is, like most of UA's interfaces, Thunderbolt, but since they tossed us lowly PC users a bone in the form of the USB Apollo Twin interface last year, maybe they'll come offer a USB Arrow. It would be a good addition to the line.
https://www.uaudio.com/audio-interfaces/arrow.html

JoeCo, maker of the line of BlackBox multi-channel capture recorders and BlueBox USB workstation interface field recorder, introduced Cello, its first desktop USB audio interface. Given the high quality and solid reliability of the Box products, this promises to be a top grade unit. It offers future-resistant recording capability at up to 384 kHz sample rate and has tweakable A/D and D/A converter filtering for optimizing performance based on sample rate and the program material.


There are two mic-only input channels with low cut filter, polarity reverse, pad, 48v phantom power, and line level inserts for outboard analog processing. There are also two ADAT optical inputs, S/PDIF input, an instrument DI with its own ¼" jack and level control, and four outputs. For studio applications, there's a built-in talkback mic and a monitor mixer. There's little technical data available at the moment, so we'll have to wait a bit for all the full specs, but it looks like a good prospect in the premium desktop audio interface category. Check the web site and there should be specs and more pictures up soon.
https://joeco.co.uk/cello-joeco-audio-interface/

Straddling the fence between a mobile and small desktop audio interface, the new MixerFace R4 from CEntrance packs a lot of functionality into a hand-sized box. The R4 has two XLR-1/4” combo input connectors for mic, line, or instruments, plus a stereo auxiliary line input. A built-in mixer with top panel controls handles the monitor mix for a zero latency (hardware) mix between inputs and DAW playback when overdubbing. The mic inputs have 48v phantom power available and there’s a switchable low-cut filter. The rig is powered by an internal rechargeable battery which they claim gives 8 hours of service on a charge. Computer interface is USB, class compliant with iOS and (some) Android phones, it can record to a Micro-SD memory card.


CEntrance might not be a familiar name, though they’ve been around for quite a while, with much of that time behind the scene. They provided drivers and license other technology to many of the manufacturers of Firewire and USB audio interfaces going back at least 15 years, and their MicPort Pro was the first plug-in USB microphone interface. Recently they’ve been building high quality portable D/A converters and headphone amplifiers.
http://www.centrance.com/products/mixerface_r4/

***Recorders and Recording Devices***

Just about when I thought handheld field recorders were dead, Roland released their new R-07. Actually, I saw this first at CES, but since I didn’t write a formal report of that show and they showed it at NAMM, I figured I’d give it a mention. The palm-sized R-07 records stereo as 16- or 24-bit WAV files at sample rates up to 96 kHz, plus several MP3 data rates.


While it’s not a true 4-track recorder, it can simultaneously record a second pair of tracks. These can be in the same format as the primary tracks but recorded at a lower level so you’ll have a shot at saving a recording that’s marred by overloads. Alternatively, you can record in both WAV and MP3 format simultaneously. There’s a pair of built-in mics and a mini phone jack for an external mic or line level input that takes precedence over the internal mics when something is plugged in. There’s a limiter that they describe as “Hybrid” where the input is continuously buffered in memory, at a level 10 or 12 dB below the primary track's level. When clipping is detected, it automatically replaces the overloaded section with the hopefully non-overloaded section in the buffer. My TASCAM handheld recorder works like that (I originally encountered this scheme in the Sony DR-50) and it's the most transparent limiter I've ever (not) heard.

The R-07 has a full remote control in the form of an iOS or Android app that talks to the recorder via Bluetooth. Furthermore, it will stream the live input to the app while recording, so not only do you have remote control including metering and record level, you're able to monitor what you're recording. I wish my TASCAM DR-44WL could do that.

Power is from two AA cells or via the USB port either from a computer or USB "charger." Recording media is a The USB port also serves as a host to copy files recorded on a micro SDHC memory card with up to 32 GB cards supported. And it comes in black, red, and white. I'll have to admit that I was a bit amused that the tone of the promo material is along the line that, sure, you can do this on your smart phone, but there's a better way – like they have to remind people that there's life beyond the phone.
https://www.roland.com/us/products/r-07/


Zoom has a couple of new field recorders. The new H1n is an update to their simplest model, the H1. The new version is fairly menu-free, with easy to read and understand shortcut buttons on the front panel for the common recording functions. Recording is to WAV files up to 24-bit 96 kHz as well as 128 to 320 kbps MP3. A limiter and low cut filter can be engaged with the press of a button. As expected, there's an X-Y pair of mics built in, with a stereo mini phone jack for an external mic or line level source.

While it's targeted toward simple, one-button recording, it's capable of doing sound-on-sound overdubs, variable speed playback, and loop playback. For audio-for-video applications, there's a tone generator for setting audio gain on a camera when feeding the camera's audio input from the H1n's headphone output. This can also serve as a slate tone. The USB port can be used for connecting an external power supply, transferring files to a computer, or the H1n can function as a USB microphone or interface for recording directly to a computer. Power comes from two AAA alkaline or rechargeable batteries that provide about 10 hours of operation (alkaline).

Zoom's new F1 is pretty much the same recorder as the H1n, repackaged to work with a camera. There are two versions, one with a belt clip and a lavaliere mic, the other with a camera mount, shock mount, and a Zoom SGH-6 shotgun mic, which can be swapped out with any of the other detachable mics for their H5 and H6 recorders.

Zooming us into the section on consoles, we have the Zoom Livetrak. It's a 12-channel mixer that's fully functional as a live mixer and can record up to 12 tracks and a stereo mix directly to a micro SD memory card. It has eight mono mic/line and two stereo line level inputs. Each channel has a 3-band EQ and a low-cut filter and effect send. There's a mute and solo button for each channel, and inputs 1 and 2 have a high impedance input suitable for an instrument DI.

There are 16 reverb/delay effects built in (effect pedals were Zoom's first product), and five individual sub-mixes for headphone feeds. There's even a talkback/slate mic and a built-in metronome. Up to nine snapshots including faders, pans, mutes, and more can be saved and recalled.
https://www.zoom-na.com/

***Mixers, Consoles, and Control Surfaces***

Sound Techniques was a home grown studio in London that grew up with bands you may have heard of like The Beatles and Rolling Stones. You couldn't buy a console off the shelf in those days, so the studio built their own, then they started building custom consoles for other studios, and formed Trident Studios. Sound Techniques, with a new crew, is now located in California and they're building consoles. The console that they showed here at NAMM is an accurate reproduction of the original Sound Techniques one that was became the well known Trident A-Range console.

The level of detail in the reproduction is really impressive. There are a lot of hand made parts – zoom in on the fuzzy photo (sorry, I can't afford a better camera on what

I make from these reports - $0) and you'll notice that the gentleman in the picture has his hand on one of the old style faders that swing in an arc, the same type used in the original console. Today's linear faders are easier to use, but it's nice that they kept a very visible bit of the console's history. This is the second one that they've built, and both have been sold. They'll happily make you one whenever you're ready. In the mean time, they're working on a more practical channel strip. Visit their web site, read the history, and dig the photos.
[http://soundtechniques.com](http://soundtechniques.com)

Looptrotter is a new brand for me but they've obviously been making friends and fans in the right places. Along with several signal processors, both full rack size and 500-series modules, they brought a modular console to the show that was custom built for producer/engineer/gearslutess Sylvia Massy. The console is, in concept, like the API Box or SSL XL-Desk but considerably more flexible and expandable - one of those "Why didn't I think of that?" things. It's a frame with bus wiring, routing, and faders, with two slots per channel that accommodate 500-series modules (theirs or any others that meet the standard), with space for some auxiliary modules which can be patched in where needed. The picture speaks for itself, I think.
[http://looptrotter.com/n_en/looptrotter-at-the-namm-show-2018-anaheim-california.html](http://looptrotter.com/n_en/looptrotter-at-the-namm-show-2018-anaheim-california.html)

Many years ago, PreSonus introduced the FaderPort, a box with a single fader and a few buttons that send MIDI control messages, allowing DAW users to do natural sounding fades the natural way. Last year they introduced the FaderPort 8, an 8-fader version. This year we have the FaderPort 16 that will make any DAW look and feel darn near like a real console – or at least a digital console - with a Select button to activate knobs and buttons for the selected channel. There's a set of transport control buttons, mute and solo buttons for each channel.

The motorized touch-sensitive 100 mm long faders feel good under the fingers and move quietly. Like the other FaderPorts, it's pre-programmed for either HUI or Mackie Control, one or both of which practically all of this generation's DAWs understand. Naturally, it has some special features that integrate tightly with PreSonus' Studio One DAW software, though it's been tested with several DAWs – Pro Tools 12, Sonar, Logic 10.4, Cubase 9.4, Harrison Mixbus, and others.
https://www.presonus.com/products/FaderPort-16

### *Signal Processors and Effects*

While it was the console (above) that caught my eye at the Looptrotter booth, it was hard to miss their new Monster compressor – well, really, it was hard to miss anything at their booth since bright yellow panels seem to be their style.


The Monster is a two-channel tube compressor that's designed with distortion in mind, or, in the vernacular, to "add character" in a mix, using a combination of compression and tube saturation. Notably, most of Looptrotter's signal processors have "sat" in their name, so the sound of tube or transformer non-linearity seems to be their thing. The Monster is a FET compressor with a characteristic curve at the onset of gain reduction (the knee) that tends to produce low order harmonics. The tube circuit adds more even order harmonics. A Mix control sums the unprocessed input signal with the processed signal the output, offering "parallel compression" within the box rather than summing within the DAW or mixing console.
http://looptrotter.com/rack_en/monster-compressor.html

Retro Instruments specializes in making updated reproductions of vintage audio gear from the era where much of our signal processing hardware came from broadcast studios of the 1950s. At this year's NAMM show, they introduced a new two-channel compressor, the Revolver. It's based on an Altec 436B single channel variable mu broadcast compressor, incorporating modifications by engineers at EMI Abbey Road Studio in the 1960s. The name comes from the fact that these compressors saw a lot of use in recordings of The Beatles. Beatles-Revolution-Revolver – get it?

The original Altec 436A had no user controls at all. The 436B added an input level control, and it wasn't until the 436C that it got attack and release controls. The most notable of the EMI modifications was the addition of a multi-position switch that delayed the start of the release after the input signal dropped below threshold. The last position of the switch was a "hold" function that kept the gain at its reduced level until it was released manually – kind of a "That's it! You ain't getting any louder" feature.

The Revolver doesn't offer the release-hold feature (Chandler makes one that does, should you want it) though Retro enhanced the original design with a low frequency side chain filter, channel linking, and a Dual Threshold control. The Dual Threshold feature is a saturation control that I suspect overdrives a tube, then adjusts the compression threshold to compensate for the increased signal level going to the gain reduction circuit.
http://retroinstruments.com/product.php?product_id=revolver

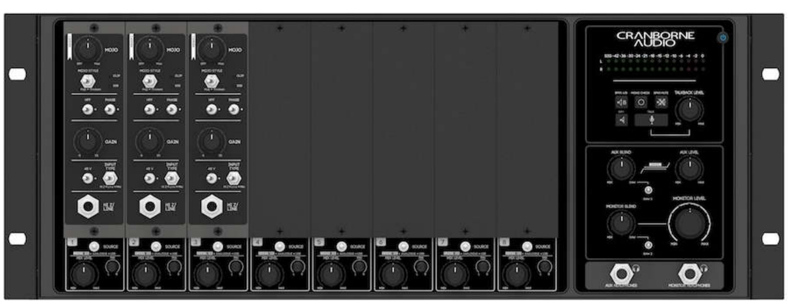
The Universal Audio Ox falls under the signal processor category, but it's a very specialized one. It's a guitar recording system that builds on the sound of your own tube amplifier. While it's electrically compatible with a solid state amp, its processing algorithms aren't calibrated to be used with one. The Ox box signal flow begins a reactive load (that is, not a simple high power resistor) that loads the amplifier in the same non-linear way that a speaker does. The loaded amplifier output is attenuated in steps from silent to sensible, and appears on a jack for connecting a speaker for monitoring the amplifier's tone, even cranked all the way up, without waking the baby or having your neighbor pound on the door,


You begin Ox setup by adjusting your amplifier to get it into the "sweet spot" for your playing style and for the song while listening, at a comfortable volume, through a speaker connected to the output of the dummy load. The properly loaded amplifier signal then goes to a processing chain inside the Ox, first to a speaker modeling algorithm, then to a mic and room modeling algorithm, and finally to an effects module which offers a 4-band EQ, 1176 compressor, delay with modulation, and a stereo plate reverb. The processing chain comes out in analog form on a pair of balanced jacks, or digitally as S/PDIF on coax or Toslink connectors. This is what you record. There's an auxiliary output that you can send the fully processed amplifier signal to a powered monitor or to the PA system, and there's also a headphone jack. The full processing chain is controlled over WiFi from either an iOS app or a Mac. This is where you select a speaker and cabinet model, choose simulated mics and position them in the room with the simulated cabinet, and adjust all the parameters of the modeled effects.



All of the gozintas and gozoutas are on the rear panel. Cleverly, they've printed the labels both in the normal orientation and upside down, so you can read them when looking down on it from its intended position on top of the amplifier. The front panel includes a volume control for the local speaker, line output, and headphone volume plus two other knobs, a Room control to add some air, and a Rig control, which selects one of six presets that you've previously set up. For something that, on the surface, appears to be a power soak and a direct box, it really goes deep into what's between the amplifier output and what you hear in the room. It'll keep you busy for a couple of months, I'll bet.
[https://www.uaudio.com/hardware/ox.html](https://www.uaudio.com/hardware/ox.html)

There are so many 500-series signal processors that I've stopped paying close attention to them, however a new rack from Cranborne Audio caught my eye. As studios add more and more of these modular processing plug-ins, inserting them into the signal chain becomes more complicated. The Cranborne R8 is an 8-space 500-series rack with a 28 in x 30 out USB audio interface and an 8-channel summing mixer with features that help to reduce the patching haywire.




The USB interface input and output count includes the 8 module slots, one S/PDIF, and two ADAT optical input and output ports. There's also word clock in and out, and MIDI in and out. With all of this connectivity and some modules, the Cranborne R8 can become the audio interface for your workstation.

Like most 500-series racks that are more than a rack and power supply, you can chain modules to create, for example, a channel strip - a mic preamp, equalizer and compressor, with its output going to the computer via USB. You can send the output of a track via USB to a module (or chain) in the rack, process it, and send the processed output back to a new track or as an insert to the original track. For mixdown, it can be used as an 8-channel summing box. There's a monitor controller section that offers switching between two speaker systems, mute, dim, and volume control, and two independent headphone outputs with adjustable balance between inputs and DAW playback. In addition, it supports an Ethernet-connected system called CAST, about which I know nothing and didn't know to ask about when I spoke to them at the show. It appears to be something unique, at least for now, to Cranborne, perhaps for remote control. Maybe we'll hear further details later when they get it finalized.
http://www.cranborne-audio.com/500r8

***Software, Plug-ins, and Apps***

In case you haven't noticed, Avid has a new scheme for version numbering. Version numbers will begin with the year and month the version is released, followed by sequential numbers for bug fixes and minor updates between real new versions. So if they come up with a new version in March 2018, that will be version 2018.3. If the next version isn't until June, there won't be a .4 or .5. I believe this is to go along with their increased frequency of updates to go along with their subscription plan.

Software plug-ins aren't really my thing, but Sonnox is a company that's always impressed me as one who comes up with new functions and processes and applies them in useful ways. Some of them can get pretty deep and pricey as well, but at this show they introduced a new series they call Toolbox. These are lower cost and easier to use tools for those who just want to get a job done. The design premise is to give high quality results with just a few controls. The first of these is a vocal processor called VoxDoubler and consists of two processes – Thicken and Widen. By golly, that's just what they do.
https://www.sonnox.com/toolbox/voxdoubler

Personal monitoring systems are making headway in many areas of live sound, from big tours, small groups playing clubs, and houses of worship. A number of today’s modestly priced mixing consoles offer remote control that offers players the ability to adjust their own monitor mixes from a mobile device. Audio Fusion Systems is a personal monitoring system that’s not tied to a specific console, but rather, broadcasts multi-channel audio via a WiFi router to a smart phone. The monitor mix is created directly on the phone via their app.

The heart of the system is a program running on a computer equipped with a multi-channel audio interface. The interface could be a digital console with a USB output that looks, to a computer, like a multi-channel interface. If your console is analog, you’ll need to add a multi-channel interface, but you can get a lot of channels for a few bucks these days, and if you’re already a multi-channel interface in your studio, you already have the hardware you need. The salient point is that that you can have remote controlled, individual monitoring mixes with any console, new or old.


There’s a fair amount of setup required initially, but most is graphically driven so it’s pretty intuitive. You can do all of the expected things like name channels, assign permissions to the individual users so one use can’t change another’s mix (or can, if you allow), customize screen layouts for the remote users, even create stage plots with channel assignments shown. This one isn’t quite hatched yet, but the developers seem to have put a lot of thought into what’s necessary and how to make it easy to use with minimal hardware expense.
[http://audiofusionsystems.com/](http://audiofusionsystems.com/)

Signal Flow (that’s a brand name) is a smart phone app for both iOS and Android that deals with the wiring part of the stage and system setup. It keeps track of sources and channel inputs, stage box numbering, sub-snakes, and, well, signal flow, like which mixer output goes to which snake input and which stage box output goes to which monitor or house speaker. You can create a stage and connection layout, print it, and share it through The Cloud with other Signal Flow users. Changes work in real time, so if the band shows up with some things that are different from their tech sheet, you can make the changes and the rest of stage crew will have the correct information. For free, you can do simple layouts

and share with one user. The more you pay (it's a monthly subscription), the more sharing you can do.
https://www.signalflowapp.com/

### *Guitar Things*

While my focus is on recording and related audio products, I found a couple of guitar things that I thought were worth including in this report.

HyVibe is an effects system for acoustic instruments, but it's not just another pedal system. Their approach is based on vibration research conducted at Ircam (the French music technology think tank) over the past seven years, and their first practical application is for a flat top acoustic guitar with built-in effects, which they were showing here.

There's a piezoelectric pickup installed under the bridge of the guitar, but you never really hear that. The pickup feeds a circuit board inside the guitar that employs DSP hardware loaded with algorithms for producing reverb, chorus, delay, and distortion effects. Effects are selected and controlled by a small control panel mounted on the upper bout of the guitar.

What differentiates the HyVibe system from your typical pedals is transducers that they install directly on the guitar top. These make the top act like a speaker cone and the effects combine with the natural sound of the guitar acoustically. It's pretty amazing to play a guitar and hear its sound with effects coming out of the sound hole and off the body. You'd think that a wooden loudspeaker would sound pretty bad, but it actually sounds quite good, which is why I chose to write about it here. One of the neatest things you can do with it, since the delay processor can store several seconds of audio, is to create a loop in real time and play along with it. With practice, you could be the Les Paul of the acoustic guitar, and if you're a street busker, you'll see some jaws drop and tips rolling in.

There're some really smart things going on with that DSP board. One is feedback suppression, necessary because the soundboard drives the pickup, which, through the transducers, drives the soundboard. One of the things that make it work well is a calibration routine that runs after the whole system is installed, which includes frequency sweeps and some pulses for phase measurements. From information recorded during the test, the processor creates a profile for that particular instrument that adjusts frequency and phase response for the most accurate reproduction.

But wait! There's more! The circuit board also contains a Bluetooth transceiver. You can play pre-recorded backing tracks from a phone in your pocket and they'll play right through your guitar – it's not the greatest Bluetooth speaker, but it doesn't suck! There's also a smart phone app for more in-depth control of the

effects than what's offered from the control panel mounted on the guitar. There's also a tuner, a metronome, EQ if you want to mess with the natural sound of the guitar, and input and output jacks. And of course if you're doing a show, you can mic the guitar with all the effects present.
[http://hyvibe.audio/](http://hyvibe.audio/)

I like mechanical things, and I like electromechanical things even better. The VSquared Vibrato System is a clever replacement for a Fender-style vibrato tailpiece that integrates the volume control with the whammy bar. What they've done is mounted a magnetic rotation sensor on the end of the shaft that goes into the body, the part that rotates so you can swing the bar out of the way of your picking hand when you don't need it.

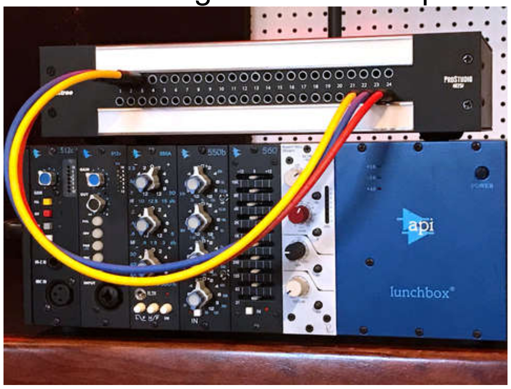
The installation kit includes both the vibrato tailpiece assembly and a replacement for the guitar's volume control that includes a pull-push switch. With the volume knob in its normal (in) position, it operates normally. With the knob pulled out, the magnetic sensor under the vibrato arm shaft replaces the volume control, allowing you do swells by swinging the whammy bar through about a 10-degree arc. You're probably wondering if, when you pull out the knob to use the VSquared volume control, you could get a surprise change in volume. I wondered the same thing as I was typing this up after the show closed, so if you're interested in one of these gadgets, be sure to ask about that.

Of course you can get the swell effect with a finger or a foot pedal volume control; this is an alternative that, at least to the inventor, seems like the motions are more coordinated when combining pitch and volume changes in the same hand. It will take some practice to get it to work for you, but it means you don't need hands like Jeff Beck's to do vibrato and swell at the same time.
[https://www.vsquaredguitar.com/](https://www.vsquaredguitar.com/)

***Other Useful Stuff***

Nearly every time someone comes to me looking for a way to not have to swap cables around in the back of the rack whenever something needs to be re-routed, my recommendation was to get a patchbay. In the old days, the console was the heart of the studio and the patchbay was nerve center, with signal routing being done with just a couple of front-accessed short cables and jacks. A DAW works well for routing interface channels to tracks, but what do you do when you have a bunch of mic preamps, some synths, and hardware effects that you don't want to give up in favor of software? A patchbay can come to the rescue.

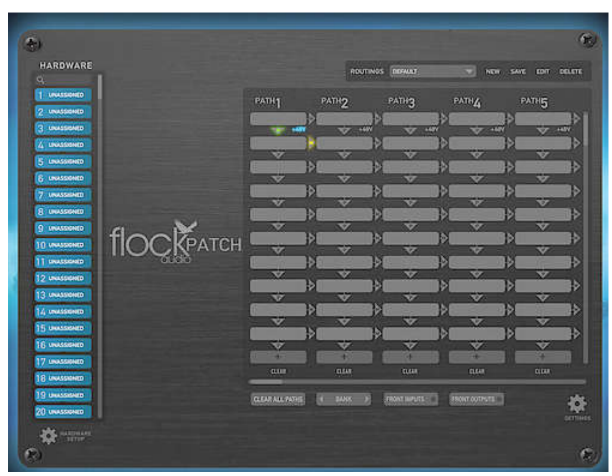
Bittree has been making patchbays for many years. They're top of the line hardware and are priced accordingly, so too many small studios have done without, chosen a less expensive patchbay that may get flaky after you've used a jack too much, or just managed with cables. This year Bittree introduced a couple of lower cost patchbays that are sized for the smaller studio. Their ProStudio 4825F is designed to sit on top of a 500-series rack like the API Lunchbox or




Radial 6-Pack. It offers two rows of 24 bantam jacks which can be set with jumpers hidden behind the label strips for full normal, half normal, or no normal connections between the top and bottom rows. Connections on the rear panel are with TASCAM-wired DB-25 connectors in groups of eight. At $625, it's not cheap, but that's about half the price of one of their full rack sized panels.

For smaller setups, the ProStudio 1625QX offers a single row of 16 jacks on the front, wired to two DB-25 connectors on the rear. This is a straightforward pass through of the connectors on your gear, providing front panel access to their jacks for easy patching between them. Off the shelf, you can get one with 16 male XLRs, 16 female XLRs, 8 of each gender, or 16 TRS jacks. It's around $300, and can save a lot of crawling around behind racks or moving boxes to get to the connector side.
[https://www.bittree.com/](https://www.bittree.com/)

For more complex studio setups, Flock Audio showed what's known as a routing switcher in the video and telecommunications business - they call it a digitally controlled patchbay. Analog signals are carried through relays, with the routing controlled by a USB-connected software application rather than by a patch cables between jacks. There are 8 TASCAM-wired DB-25 connectors on the rear panel for connecting 32 inputs and 32 outputs, plus two XLR-1/4" combo connectors on the front panel to accommodate "visitors."




The application is configured by making a list of all the sources and destinations, with routing done simply by dragging an output or input from the hardware list on to a path. You can send an output to multiple devices, though it's not a mixer so you can't drag multiple outputs to one input. The app won't let you.

While many of us shy away from using a traditional patchbay for microphones (there are a few good reasons why) the Flock Patch has a 48 volt phantom power supply so mics can be powered directly from the patchbay without worrying about “hot patching” when power is supplied by a mic preamp. Phantom power can be locked out for mics that you don’t want to power, and there are safety locks to prevent phantom powering the output of a hardware device.

Patching setups can be saved and recalled, so frequently used routings that you’d set up using half- or full-normalled jack pairs can be recalled with a few clicks. If, for example, you have a favorite vocal tracking chain with a specific preamp, compressor, equalizer and limiter, or a favorite mastering chain, these can be saved by name and recalled when you need it. Internal wiring is done with cable with 110Ω characteristic impedance, so it’s good excellent for patching AES/EBU digital connections. It’s priced out of the range of most personal studios, but at around $3,000, it’s close to the cost of a mechanical patchbay with the same number of patch points and enough cables to patch everything you’d need, and you’ll never have to clean another jack.
https://www.flocktechnologies.com

### *Last But Not Least – Education*

Diagonally across an aisle from where Bose Professional was exhibiting speaker systems was BOSEbuild. This isn’t another loudspeaker system, but rather the first of a series of science educational programs, based, as might be expected from Bose, around the loudspeaker. Lessons in this series begin with electromagnetism, going forward through the principles of sound and hearing, frequency, amplitude, and resonance. The principles are illustrated throughout with experiments using an iOS mobile device and the supplied parts kit.


It’s appropriate for children from about age 8, and it’s similar to the things that I had fun with when I was a kid. I wound wire around a nail to make an electromagnet, whereas in the Bose kit, you get a permanent magnet and a bobbin with wire wrapped around it that slips over the magnet. First, you make the bobbin jump when a pulse is applied to it through the circuit board connected to your iPad, then you watch it move slowly up and down in response to a low frequency alternating current. Kit parts are assembled as the lessons progress, ultimately becoming a functional Bluetooth loudspeaker. Along the way, there are graphic and aural examples of sounds and how they combine to make music.

Lessons can be parent-guided, self-guided, and there's a classroom curriculum with bulk pricing for schools. This is really more of a CES than a NAMM product (maybe it was there, too, and I missed it) but I'm glad they brought it out, if only to remind us that music-related education is an important function of NAMM. Retail price is $149, about right for a gift for a kid with a bit of scientific curiosity and a love of music.
https://build.bose.com/

And that's all, folks. See you next show.