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ASSISTED LIVING

We tend to overestimate the effect of a technology in the short run and underestimate the effect in the long run. Not my words, but those of the scientist Roy Charles Amara — and a pretty good summing up of the history of music tech. When sampling became affordable in the early '80s, we were told it meant the death of orchestras and the end of human musicianship. Four decades on, there's no doubt that sampling has had profound effects, but they're more subtle and more varied than anyone could have predicted.

Today's doom-mongers and tech evangelists are busy making extravagant claims about AI. And we're already seeing that AI models can produce acceptable pastiches of human-created art, not only in music but also in the visual arts and literature. But in terms of society as a whole, the extreme predictions of both optimists and pessimists remain unfulfilled.

Billions have been invested in companies that still have no path to becoming profitable. Issues such as hallucination are being waved away, even though they may represent fundamental problems with the technology. GPU makers are lending money to AI companies to buy more GPUs. From where I stand, it looks like a bubble ready to burst. But that's not to say that AI won't be transformative in the long term.

It's just that the transformations it actually brings might not be the ones we expect.

The dotcom crash around the turn of the century didn't lead to the Internet being switched off, or the end of e-commerce as a concept. Instead, it culled thousands of lame startups to leave the field clear for those with a robust business model. But coding websites wasn't hard, and once coded, they cost little to run. That's very much not the case with generative AI, which requires colossal effort to implement, and consumes enormous resources.

In the near term, my feeling is that the bigger transformation will be the quiet one wrought by assistive AI. And SoundFlow 6, as reviewed in this issue, is a great example. Its AI features are not about eliminating humans from the creative process, but enabling humans to learn faster and spend less time doing repetitive tasks. Personally, I've always found video tutorials to be frustrating, and we all know how hard it can be to find the information you need in a 300-page PDF manual. The potential for an AI trained by experts on your chosen DAW to deliver the answers you need fast, and carry out routine session management jobs, seems to me genuinely exciting. Hey, SoundFlow! Colour all the drum tracks purple and route the tom mics to a bus, please. And put the kettle on. **///**

Sam Inglis
Editor In Chief

**"Today's
doom-mongers and
tech evangelists
are busy making
extravagant
claims about AI."**



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**COMPLETE
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\$20,000

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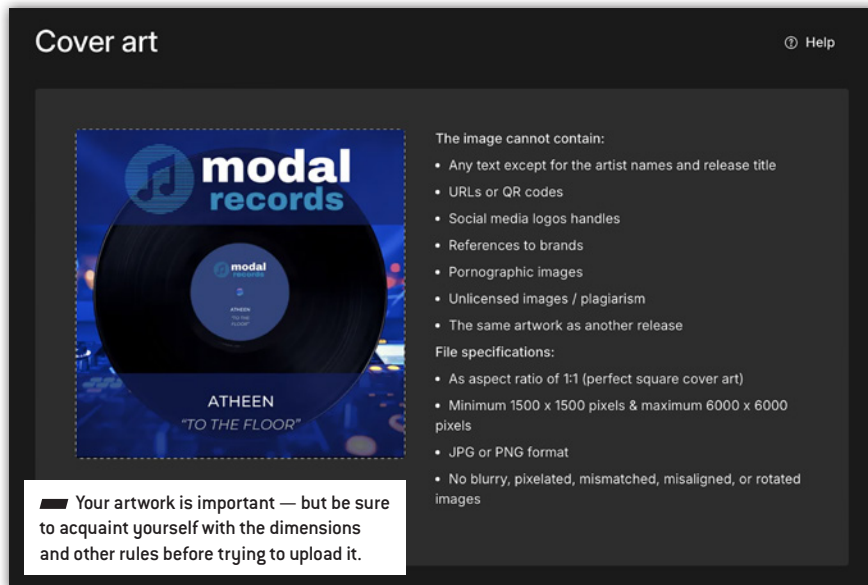
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With our nu-disco tune mixed and mastered, we turn our attention to distributing and promoting it...



THE SOS TEAM

In previous articles, we've described using SOS For Artists (SOSFA) to create, mix and master a nu-disco track. Now for the final steps: distribution and promotion. There's no single 'right' way to promote a tune, but while most independent artists won't want to spend big bucks, following some common industry practices might just boost a track's 'discoverability'. Before all that, though, we need to give our song a name! After brainstorming around 30 potential titles, 'To The Floor' emerged as our favourite, and also inspired a topline idea that resulted in a vocal version.

The next step was to gather together all the assets necessary for the release campaign. Alongside the audio, you'll usually need graphics, videos and written content. Typically, it's best to create video content first, and then take screenshots for your artwork and promotional visuals. But not everyone will have the skills, budget or time to produce a traditional music video; more on this below. In this case, we decided to focus first on the artwork, and then to produce a simple 'artwork video' based on it.

Release Artwork

For SOSFA, the minimum spec for your main artwork image is 1500 x 1500 pixels. We'd recommend at least 3000 x 3000 pixels — that will look good if anyone needs to print it at 300dpi, and is of sufficient

quality that you're 'future-proofed'. The maximum supported size is 6000 x 6000 pixels, and any image must adhere to this 1:1 aspect ratio (in other words, it must be square). If you're 'borrowing' images from elsewhere, then do be sure to check the usage rights. You should keep any text on the image to a minimum; only the artist and track name are allowed. And you can export a smaller version for use with track listings and online promotion.

You'll need to use a third-party tool to create your artwork, and while it's possible to do this with various phone apps now, if you want to take your graphics more seriously then Photoshop, Canva and

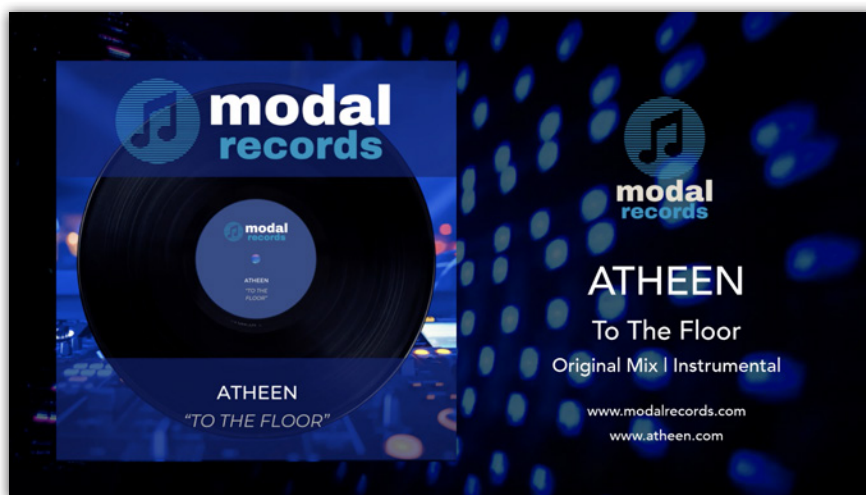
Affinity Photo are all capable and popular tools. While you're focused on designing your graphics, consider also creating some themed headers that you can use to freshen up your website and social profiles on release day. And with an eye on the later promotion phase, you could create other social graphics now too — some imagery promoting merchandise, tour dates and launch parties, for example.

Simple Videos

SOSFA's focus is on distributing your music, not videos but, like it or not, videos are a necessity when promoting your music today. Thankfully, creating them needn't be daunting.

Those who are comfortable on camera can shoot their own footage and turn it into a music video. Or you might simply film a live performance. You could also hire freelancers or connect with fellow creators on the SOS Forum to fill any skill gaps (after all, building a strong team is key to any successful endeavour!). But if performing on camera or motion graphics aren't your strength and you want to go it alone, then simple lyric videos could be a good option. These remain a popular choice, and you can use a basic video editor to overlay lyrics onto background images, just adding fades or simple animations between them. Aligning text on the timeline can take a little time, but it's a great opportunity to get creative with fonts and visuals to tell a story.

Since we're releasing a nu-disco track, we thought a glitterball-themed promo video the perfect choice! To create it, we downloaded a royalty-free seamless loop from our own Canva subscription (www.canva.com). You can buy Canva's stock



■ While SOSFA focuses on the audio side, videos can be an important promotional tool — and if you've not made one, creating something useful based on your artwork might be less onerous than you imagine!

1	ISRC Code	Year	Track	Album	Mix	Release Date	Artist	Credits	Catalogue N°	Barcode	BPM	Key	Duration	Genre	Description	Keywords
378	UK-M3L-25-00082	2025	To The Floor	To The Floor	Main Mix	24/10/25	Atheen	n/a	ibc	ibc	124 C minor	03:41	Nu Disco, Funky House	A lively dancefloor release from lively, fun, dance, uplifting, nu disco, funk, acid		
379	UK-M3L-25-00083	2025	To The Floor	To The Floor	Instrumental	24/10/25	Atheen	n/a	ibc	ibc	124 C minor	03:41	Nu Disco, Funky House	A lively dancefloor release from lively, fun, dance, uplifting, nu disco, funk, acid		
380	UK-M3L-25-00084	2025	To The Floor	To The Floor	STEM - Bass	24/10/25	Atheen	n/a	ibc	ibc	124 C minor	03:41	Nu Disco, Funky House	A lively dancefloor release from lively, fun, dance, uplifting, nu disco, funk, acid		
381	UK-M3L-25-00085	2025	To The Floor	To The Floor	STEM - BV's	24/10/25	Atheen	n/a	ibc	ibc	124 C minor	03:41	Nu Disco, Funky House	A lively dancefloor release from lively, fun, dance, uplifting, nu disco, funk, acid		
382	UK-M3L-25-00086	2025	To The Floor	To The Floor	STEM - Drums	24/10/25	Atheen	n/a	ibc	ibc	124 C minor	03:41	Nu Disco, Funky House	A lively dancefloor release from lively, fun, dance, uplifting, nu disco, funk, acid		
383	UK-M3L-25-00087	2025	To The Floor	To The Floor	STEM - Effects	24/10/25	Atheen	n/a	ibc	ibc	124 C minor	03:41	Nu Disco, Funky House	A lively dancefloor release from lively, fun, dance, uplifting, nu disco, funk, acid		
384	UK-M3L-25-00088	2025	To The Floor	To The Floor	STEM - Guitar	24/10/25	Atheen	n/a	ibc	ibc	124 C minor	03:41	Nu Disco, Funky House	A lively dancefloor release from lively, fun, dance, uplifting, nu disco, funk, acid		
385	UK-M3L-25-00089	2025	To The Floor	To The Floor	STEM - Keys	24/10/25	Atheen	n/a	ibc	ibc	124 C minor	03:41	Nu Disco, Funky House	A lively dancefloor release from lively, fun, dance, uplifting, nu disco, funk, acid		
386	UK-M3L-25-00090	2025	To The Floor	To The Floor	STEM - Strings	24/10/25	Atheen	n/a	ibc	ibc	124 C minor	03:41	Nu Disco, Funky House	A lively dancefloor release from lively, fun, dance, uplifting, nu disco, funk, acid		
387	UK-M3L-25-00091	2025	To The Floor	To The Floor	STEM - Vocal	24/10/25	Atheen	n/a	ibc	ibc	124 C minor	03:41	Nu Disco, Funky House	A lively dancefloor release from lively, fun, dance, uplifting, nu disco, funk, acid		

It's a good idea to get all your metadata together in advance, in a spreadsheet...

Metadatics											
General											
Track Title	To The Floor										
Track Artist	Modal Publishing										
Album Artist	lively, fun, dance, uplifting, nu disco, funk, acid jazz, cheerfu										
Composer	Atheen Atheen Spencer (100%) (PRS CAE 703165868)										
Work											
Year	2025										
Track Number	1 of 2										
Disc Number											
File Name	Track Title	Track Artist	Album Artist	Composer	Album	BPM	Grouping				
To The Floor - STEM - Bass.aif	To The Floor - STEM - bass	Modal Publishing	lively, fun, dance,...	Atheen Atheen...	To The Floor	124	Modal Publishing (one-stop)...				
To The Floor - STEM - Drums.aif	To The Floor - STEM - drums	Modal Publishing	lively, fun, dance,...	Atheen Atheen...	To The Floor	124	Modal Publishing (one-stop)...				
To The Floor - STEM - Effects.aif	To The Floor - STEM - effects	Modal Publishing	lively, fun, dance,...	Atheen Atheen...	To The Floor	124	Modal Publishing (one-stop)...				
To The Floor - STEM - Guitar.aif	To The Floor - STEM - guitar	Modal Publishing	lively, fun, dance,...	Atheen Atheen...	To The Floor	124	Modal Publishing (one-stop)...				
To The Floor - STEM - Keys.aif	To The Floor - STEM - keys	Modal Publishing	lively, fun, dance,...	Atheen Atheen...	To The Floor	124	Modal Publishing (one-stop)...				
To The Floor - STEM - Strings.aif	To The Floor - STEM - strings	Modal Publishing	lively, fun, dance,...	Atheen Atheen...	To The Floor	124	Modal Publishing (one-stop)...				
To The Floor (Instrumental).aif	To The Floor (Instrumental)	Modal Publishing	lively, fun, dance,...	Atheen Atheen...	To The Floor	124	Modal Publishing (one-stop)...				
To The Floor.aif	To The Floor	Modal Publishing	lively, fun, dance,...	Atheen Atheen...	To The Floor	124	Modal Publishing (one-stop)...				

...and you can then manage it in a third-party app such as Metadatics. Note that you'll need metadata for each version you release, including any stems intended for sync.

footage for a modest fee if you're not a subscriber, but there are lots of other places where you can source royalty-free footage too. We then edited the video in DaVinci Resolve (there's a free version if you're new to video editing: www.blackmagicdesign.com/uk/products/davinciresolve), layering artwork and text on top, to create our simple animated visual.

With a 16:9 (widescreen landscape) video ready for YouTube, and a shorter Facebook edit, a 9:16 (portrait, for phones) version was created for Instagram, YouTube Shorts and Spotify Canvas (the looping visuals that accompany some Spotify releases). You can't cover every platform, but it makes good sense to focus on the few platforms where your audience is likely to be most active, and these cover a lot of bases.

The Boring Part!

With the visuals in the bag, it's time to don your 'record label' hat. Start by organising your track's metadata, ideally in a spreadsheet like Excel, with columns for ISRC, track and product titles, mix version, release date, credits, duration, catalogue number, bpm, key, genre, description, and keywords. Not all of that is strictly required for release, but it will make promotion and uploads to sync platforms much easier.

Let's look at ISRC codes first. This is an international identifier for your tracks, and typically it's assigned automatically during distribution. If you're running your own label, you can obtain a unique three-digit identifier and country code through PPL in the UK (if you're elsewhere you can find your local Performing Rights Organisation at: <https://isrc.ifpi.org/en/>

[contact/national-agency-contacts](https://www.contactnationalagency-contacts)). The code combines the country abbreviation, identifier, year, and a five-digit track number; for example 'UK-M3L-25-00082'. Be sure to assign a separate ISRC for every mix of each track, including each stem group that you'll be uploading for sync.

In the description column, you should include a brief sentence or two about the track's style. For keywords, add 15-20 terms reflecting its mood. For example: 'upbeat', 'lively', 'sad', or 'ominous'. Catalogue and barcode numbers are generated automatically by the SOSFA platform, so these fields can be filled in later.

Next, add your metadata to the tracks, starting with the release audio. There are lots of dedicated tagging tools that can make this straightforward; we used the Mac-only Metadatics (www.markvapps.com/metadatics) and entered the Title, Artist, Album (you could put a single's name here), Genre, Year, and bpm. For sync purposes, duplicate your audio folder. In your metadata program, add your full name and PRO registration number in the Composer field and, if you are the sole rightsholder and can offer instant clearance, enter 'One Stop' in the Group field. Include a contact email address here as well. Use the comments

»

Artists
[How to make sure artist profiles link properly?](#)

Primary artists ?

Atheen
⊕
×

Add artist...

Featured artists (optional) ?

Add artist...

Remixers (optional)

Add artist...

Contributors (optional)

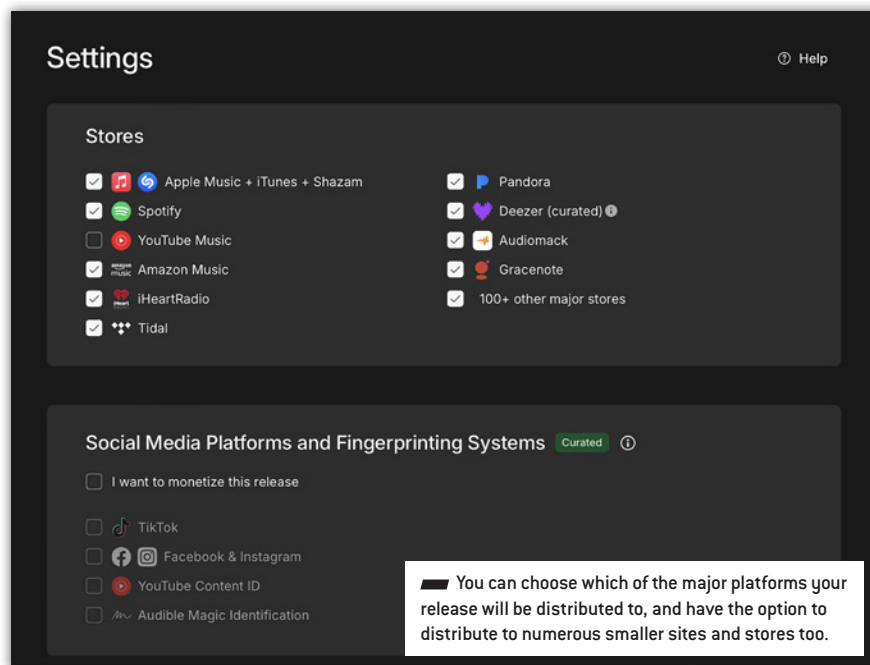
+ Add

Producers ?

Atheen
×

×
⌵

With all your data prepped, and your campaign materials ready, uploading your release to SOSFA is easy — and you'll be given a friendly nudge if you miss any important information.



» box for a brief track description, website, contact info, ISRC, Key, and Time Signature. Finally, paste in your comma-separated keywords — the album artist field works well for this, since it's searchable.

Electronic Press Kit & Press Release

Traditional physical media packs have largely been replaced by Electronic Press Kits (EPK), hosted on your own website or platforms like DISCO (www.disco.ac), which seems to be gaining popularity. Make sure it includes your artist name, promotional and release images, track listing, release date, credits, biography, contact info, target markets and a list of a few well-known artists with a similar sound. Include several high-res professional images so that journalists can easily find what they need.

You'll also need a press release, which sounds grand but is super-easy! We'd suggest creating a one-page A4 Word document and including your contact details and a logo at the top. Begin with 'FOR IMMEDIATE RELEASE', add your promotional text, and end with '###' or ENDS to signal the end of the release. The release should briefly outline your achievements and highlight your latest release, but do try to write something original and engaging to attract attention!

Distribution & Registration

With all materials in place, it's time for distribution and registration. A release date 4-8 weeks ahead is ideal, but this track

had a faster turnaround, and SOSFA can get releases live in as little as 2-7 days if your profile is already set up — though allowing more time obviously helps with pre-release promotion.

In SOSFA, go to the Distribution tab and select 'Add Release.' With all assets and metadata ready, uploading is straightforward. Click into each track to enter title, artist, mix version and other details. The platform will automatically flag any missing information. On the Release Info tab, add yourself as the composition owner, and either you or your label name as the master recording owner, according to your PPL registration. The Cover Art tab lists artwork guidelines as a helpful reminder.

The Release Settings tab lets you choose distribution platforms, set your release date, and opt for SOSFA to collect YouTube royalties for you — useful if your channel isn't yet monetised. The final page allows you to review all information before submitting. You can save and return later for a final check if you prefer — mistakes can be corrected after submission, but updates may take a couple of weeks to appear.

Once your track is approved for distribution, register it with your local Performing Rights Organisation (PRO) to collect royalties. Here in the UK, this is the Performing Rights Society, or PRS (www.prsformusic.com). Membership requires ID and a small fee, but it should pay off if you are actively promoting your music to radio and submitting your live performance information. Once registered,

adding your songs is a quick process, especially if you use the metadata database we discussed above.

If you're running a label, you'll also need to register your release with the organisation that issued your ISRC code, in our case PPL UK. Registration is free and you can set up a rightsholder account as an individual or under your label if registered as a company. Uploading is straightforward: register individual tracks first, then create a separate registration for the overall release, in this case a single with two mixes. It's advisable to wait for PRS registration confirmation first, since you can link the two databases by adding in the ISWC (International Standard Musical Work Code) provided by PRS.

Finally, add your release to MusicBrainz (<https://musicbrainz.org>) for industry visibility and use Discogs (www.discogs.com) to give even more visual presence, particularly if you're releasing physical formats too.

The Release Campaign

By now, you might be getting an idea of how good record labels actually earn their share of the income! While there's still more legwork to come, it's perfectly normal to start small with your first release campaign, and increase your efforts with each new release, as you acquire more contacts and followers.

There are three main groups on whom, ideally, you need to target your promotion efforts:

- The media — under which heading we'd include DJs, radio stations, blogs, magazines, reviewers, tastemakers, playlists, venues, and so on.
- Music supervisors for sync.
- Your fanbase.

Note that your focus should be on building long-term, mutually beneficial relationships. To that end, if you're taking your music career seriously, consider hiring a promoter or enlisting a friend to help ease the workload. But even without that, you can start building your contact list by searching locally and online for industry professionals in your genre. Use Google, LinkedIn, and attend events with business cards to grow your network. Contact each person individually where possible, sharing a brief introduction and a link to your press kit, and offer the option to join a mailing list (never add contacts automatically; in many countries it's illegal, and in any case you

want people to notice you, not report you as a spammer!). Make sure the tracks on your EPK are actually downloadable.

Emerging artists can upload their track to BBC Introducing (www.bbc.co.uk/introducing) for potential local radio coverage and pursue interviews with local and national publications, TV, radio and online outlets. Try to arrange interviews to coincide with gig dates in that area. You can use the SOSFA Promotion tab to set up your official Spotify for Artists account, where you can apply to be featured on Spotify playlists. You can also sign up to Musixmatch here to upload lyrics, and further expand your visibility.

When it comes to music supervisors, you'll need to reach out individually, with a polite message and a link to your tracks. Consider uploading your releases to Songtradr (www.songtradr.com), where they can be licensed for sync immediately. Other platforms exist, of course, but note that many of them don't pay royalties! Alternatively, a publisher can seek out sync placements in exchange for a share of the licensing and royalty fees, so that

you're not having to solely handle any legal paperwork.

Finally, don't forget to set up a campaign to promote your release to fans, including as a minimum scheduled social posts and a dedicated web page. Consider running launch events (online, offline or both) and dedicated gigs. Take note of your more enthusiastic fans and encourage them to share links, and make sure you share any press coverage that you receive. Some artists will even stage local publicity stunts to gain further exposure, but we'll leave those ideas to you!

Ongoing Promotion

Whatever you do, don't be tempted to view 'release day' as the end of the campaign. You need to generate momentum, so continue updating banners and promoting to both fans and the industry for at least a couple of weeks beyond the release date. Specifically on SOSFA, you can go to your release page to access store links — these can be easily shared — and can visit the Tools tab to link your official YouTube channel,

to ensure that all content on the platform is earning income for you.

A smart way to boost streams is creating a themed playlist around your release, including popular or similar artists. Give it a clear, appealing title, such as '8 Hours of Relaxing Spa Music' or 'Epic Gaming Soundtrack,' that can attract both listeners and businesses who use Spotify. Tracks can then often be discovered in venues like restaurants, coffee shops and gyms via Shazam... Finally, you can extend a track's life even further by performing, sharing videos and even releasing remixes in the weeks following the initial release.

Rinse & Repeat

As you can see, running a release campaign requires a heck of a lot of admin and contact building, and this quick guide only really scratches the surface. You could also explore collaborations, cross-promotions with other artists and paid campaigns on platforms like Google, Facebook and Spotify... and much more. But hopefully we've given you some ideas. Whatever approach you take, have fun with it! **///**



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Focal Lensys Pro

Closed-back Headphones

Focal's latest studio-focused headphones aim to deliver reference-quality monitoring in a closed-back format.

SAM INGLIS

When it comes to headphones and loudspeakers, the divide between studio and hi-fi models is sometimes more apparent than real. There are plenty of hi-fi enthusiasts who say they want to hear music on the same type of system that was used to mix it, or at least on a flat-sounding playback rig. So although it's not uncommon for a manufacturer to use different distributors, retailers and publicity materials for the two markets, it's also not uncommon for the products themselves to be pretty similar.

Visit the website of French monitoring maestros Focal, for example, and you'll find their headphone range divided into High-Fidelity, Wireless and Professional categories. But it doesn't take Sherlock Holmes to figure out that they're all built on the same basic chassis and technology, presumably with voicing tweaks to accommodate listener preferences. The Clear MG Professional open-back headphones I reviewed in *SOS* June 2021 have a direct counterpart in the hi-fi world, while the subject of this review also bears a strong resemblance to existing hi-fi models from Focal.

Lens Flair

The Lensys Pro is a new closed-back headphone model that, as far as I can tell, uses the same housing as the more hi-fi-oriented Azurys model. This means that the earcups actually resemble those of an open-back design, with an attractive arrangement of holes laid out in four concentric rings. Behind these holes, however, what looks like a fine gauze is actually a layer of rigid material that blocks sound from entering and leaving, and the degree of acoustic isolation on offer is pretty good.

The earcups themselves are of an elongated oval in shape and, as seems to be the fashion these days, are tilted so that the lower half sits forward of the top. The yokes offer plenty of rotation on both the fore-and-aft and the up-and-down axes, and they are mounted on notched metal strips that slide in and out of the headband to accommodate different head sizes. The headband itself is padded, but rather narrow, and its pressure on the top of the head does become noticeable after a while. In general, though, these are pretty comfortable headphones, and at 306g are relatively lightweight. The foam earpads are a little deeper than most, and are field-replaceable.

The Lensys Pros connect to the outside world through a mini-jack socket in the



base of the left earcup. Straight and curly cables are supplied, as is a good-quality semi-rigid case. Internally, the Lensys Pros use Focal's proprietary aluminium-magnesium 'M-profile' moving-coil drivers, and Focal say that "their acoustic load has been optimised for operation similar to that of

a bass-reflex loudspeaker".

They present a nominal impedance of 26Ω and a sensitivity of 100dB SPL for a 1mW signal at 1kHz. Refreshingly, Focal give a frequency response with tolerances, in this case 5Hz to 22kHz, ±3dB. Distortion is given as 0.3% for a 1kHz signal at 100dB SPL; this is decent, though it would be nice to know what the figure is at 100Hz too.

Focal Points

Although they're about half the price of the Clear MG Professional, the Lensys

»



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DESIGN

Although the outer shell of the earcups is perforated, a second layer of rigid material within prevents sound from entering or leaving.



» Pros are still relatively costly for a pair of closed-back headphones. You won't be buying these in sets of five to hand out to musicians on the live-room floor. So, if you are considering them for studio roles, you'd want them to be good enough to use for tasks that demand critical listening: editing, sound design and mixing, for example. Are they?

In my view, yes they are; and, moreover, they have some subjective qualities that you don't find in too many headphones. The tonal balance is



The Lensys Pros are shipped in a good-quality fitted case.

broadly neutral, but inasmuch as it departs from the flat, the most noticeable feature is not the presence boost that is typical of so many closed-back headphones. Rather, the Lensys Pros slightly bring forward the midrange, with a broad emphasis that is most prominent around 300Hz but apparent right up to 1kHz or so. There's a subtle yet noticeable 'thickness' to the sound that tends to make other headphones sound 'scooped' when you switch away from them.

As far as their utility goes, I think this is mostly

a good thing, especially for mixing. In fact, I fancied that the Lensys Pro had just a little of the character of my Yamaha NS10s, in that they are brutally revealing of boxiness, honkiness, unwanted resonances and general low-mid clutter. The flip side of this is that if you mix only on the Lensys Pro, I think you'd need to take care not to clean things up to the point where the mix becomes thin-sounding on other systems, but I'm sure you'd get used to this fairly quickly. And, of course,

“There's a subtle yet noticeable 'thickness' to the sound that tends to make other headphones sound 'scooped' when you switch away from them.”

the Lensys Pros don't suffer from the NS10's bandwidth restrictions at either end. Sibilants, cymbal crashes and other high-frequency events are represented naturally and smoothly without any hype, which means the Lensys Pros don't become fatiguing during extended use. Nor are they plagued by the artificial boost in the 100-150 Hz region that some cheaper cans use to compensate for a lack of true bass.

Although the Lensys Pros are not cheap, then, they do offer a sound character that is relatively unusual. Not having heard the Azurys model with which they clearly share much DNA, I don't know if this is unique to the Lensys Pro, but it's a voicing I don't associate with hi-fi products, which tend to have a flattering, 'smile curve' presentation. And, to my mind, it's one that is particularly useful for studio tasks. Auditioning a few of my own mixes through the Lensys Pros often had me reaching for the EQ, and the results were usually the better for it! **///**

summary

By gently emphasising the all-important midrange, Focal have given the Lensys Pro a valuable USP for mixing applications.

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Atomic Analog Tonograf EQ

A vintage German console EQ gets a new lease of life in 500-series format.

NEIL ROGERS

Vinny Wood is based in Iceland, and there, under the name Atomic Analog, he repairs high-quality vintage analogue pro audio equipment and also makes his own. Back in *SOS* October 2024, Sam Inglis reviewed his debut release, a substantial RCA OP-6-based preamp called the Special OP-6 (<https://sosm.ag/aa-special-op6>) and was impressed. Now, Atomic have released a 500-series equaliser called the Tonograf EQ. It may be smaller than the OP-6, but there's ample evidence that Vinny has taken the same forensic approach to recreating another obscure but desirable bit of kit. Atomic kindly made available a pair of Tonograf's for review, and that was great because it's nice to have the option to use them on stereo sources when recording, and also I often enjoy running a stereo mix through analogue EQ.

Picking Up The TAB

The Tonograf EQ is billed as a part-for-part recreation of the TAB W95 equaliser module. Developed in the early 1960s, the W95 was a transistor-based update of earlier valve-based designs, and, like most such vintage studio technology, it commands high prices today — apparently with good reason. On first hearing it, Vinny says he was immediately taken by the sonic effect of the discrete germanium circuit, which contains both inductors and transformers. So much so, in fact, that it left him with a “blissful impression” even when no EQ boosts or cuts were applied.

The R&D involved in recreating this device was a challenge for a one-man boutique company, not least because many of the original components are simply not available now. It involved detailed analysis of the components, in the case of the inductors so that custom recreations could be commissioned, and the swapping of ‘surrogate’ parts into his original TAB unit until Vinny was satisfied that they behaved exactly as the originals did. The resulting design features Lundahl input and output transformers, those custom-made inductors and ‘name’ components such as Panasonic, AVX and Vishay capacitors and Vishay and Yageo resistors, chosen not only to match the characteristics of the original without compromise, but also to ensure this new product endures as long as the originals have.

Working within the tight confines of the 500-series format, Vinny opted not to adhere to the aesthetics of the original units, but wanted to retain the ‘tactile’ feel of their detented controls and ensure that all the same simple EQ options were available. Accordingly, you can find controls for low and high shelving capacitive EQ bands in the bottom half of the control panel, and an inductor-based proportional-Q mid band in the upper half. There are some welcome additions to the original design, though. First, while the high shelf of the original TAB module was fixed at 10kHz, here you have the option of setting it to 7, 10, or 16 kHz, giving the user some useful flexibility. The low shelf remains set at 60Hz, and both shelves can cut or boost by up to 12dB. Second, the mid band on the original modules was a boost-only affair, and in the Tonograf the design has been modified to allow the mid band controls to apply cuts too ($\pm 9\text{dB}$), as well as adding another centre-frequency option for good measure: you have 500 and 700 Hz, and 1, 1.4, 2 and 3 kHz to choose from.

The nominal operating level is +4dBu and you can find other specifications on the Atomic Audio website but, really, noise and distortion specs are almost an irrelevance for characterful vintage-inspired designs such as this! The modules need a host capable of



supplying each one with 180mA, which might be a little higher than the official VPR Alliance spec permits but it is within the capability of pretty much all modern 500-series racks.

On Test

Having installed them in my IGS Panzer rack, I decided to see what the Tonograf's could offer on the project I was working on at the time, so patched them into the stereo mix bus for some quick rough mixes I needed to send to my client. I began playing with the low- and high-shelf filters, and was immediately taken with the way a 16kHz boost seemed to open up this piano-led track. I was also impressed at the extra weight that the low shelf was giving me.

I went on to use the Tonograf's on other mixes, and have to say that I really liked them for mix-bus EQ'ing. The two units seemed to be very well matched

»



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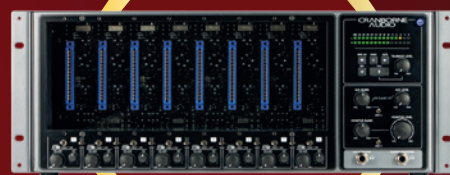
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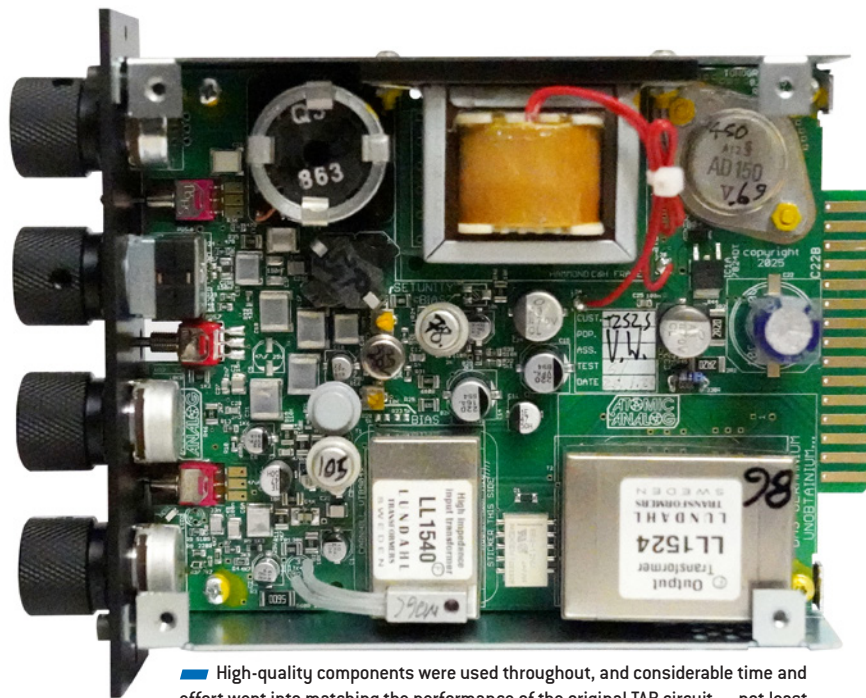
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» and suitably 'stable' when used on stereo sources, and the sweetening effect of these circuits — a subtle but pleasing sort of 'sheen', combined with that satisfying touch of fullness you get from transformer-based designs — as well as the broad-strokes EQ options should be very appealing to engineers as they approach the final stages of a mix or master.

My first proper tracking session with these units was on a piano session. It's an instrument for which I've developed a few mic techniques over the years to make my studio's somewhat dark-sounding upright sound richer and, well, generally a bit more 'grand'. The Tonografs' low and high shelves helped a great deal, with both set to boost at just shy of the 3 o'clock position, but while the EQ curve was obviously helping, there was more to it. Switching the EQs in and out revealed just what a dramatic effect they were having, and my client remarked on how good the piano was sounding.

The rest of my time with these EQ modules coincided with a very busy period at my studio, during which I recorded a series of young guitar bands who were chasing a sound that was very familiar to a '90s kid like me! The Tonografs sounded superb on drum close mics, and the 60Hz shelving filter in particular worked nicely for adding extra weight on kick drums; it was also great to be able to cut some boxy frequencies at 500Hz or add more presence if needed. When paired with the classic close snare mic option of an SM57, it was possible to get heavy-handed with the midrange boost at 1.4 or 3 kHz, and I found myself pairing this with a high shelf boost set at 7kHz or a cut at one of the higher frequency options. In both these high-SPL settings, I quickly developed more of a sense of the 'colour' being imparted by the electronics over and above the boosts and cuts; it was easy to hear how the subtle amount of saturation complemented the music. I had a similarly positive experience using these EQs on close-miked



High-quality components were used throughout, and considerable time and effort went into matching the performance of the original TAB circuit — not least the mid band's inductor, which had to be analysed and recreated.

guitar cabinets: they worked a treat in those in-depth 'tone-chasing' sessions, where you just want to be able to lean over and pull a little brightness up in the sound or add more weight.

Summing Up

As you can probably tell, I was rather taken with these EQs! They're the kind of thing that you find yourself wanting to leave in your recording chain whenever possible, even if you're not actually making any EQ moves. That's precisely how I found myself using them on vocals, in fact — sitting there as a sweetener, they're gently flattering all the time but, should you need it, they also offer simple tone-shaping at the twist of a knob.

Of course, a diligent reviewer must also search for negatives. But while I did

find a couple of points worth noting, it's not a significant list in the grand scheme of things. For instance, I often had to get out of my chair to see what settings I'd dialled in, but this is a common issue for me with 500-series modules. Also worth noting is that these units do get very hot

when they've been on a while, and the manual rightly suggests you give some thought to where to place them in your 500-series chassis.

Finally, boutique tools such as these rarely come cheap, and certainly these are at the more expensive end of what's available for the 500-series format. Having said that, these are some of the nicest-sounding 500-series EQs that I've used, and if you stuck two of these modules in a 19-inch rackmount chassis for a similar price, they'd be competing with products that cost much more. But of course these were originally console EQs, so I think it makes good sense to stick with a modular format — there are still mixers that can host 500-series modules like this, after all!

The bottom line is that if you're carefully assembling a high-end recording and mixing setup, these would be a fantastic addition, and are well worth checking out. ■■■

summary

The classy-looking Tonograf EQ packs a lot of vintage analogue muscle into a single 500-series module. They combine a subtle but pleasing sound character with a simple-to-use EQ that works superbly for all-round recording and mixing duties.

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The Tonograf has more features, yet is more compact than the W95C.

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NICK ROTHWELL

Over the last few years I've been seeing an increase in software bloat in audio plug-ins, whether that's the hundreds of presets, 3D animated graphics or just the complexity of hybrid synthesis models with long chains of built-in effects. It's therefore a relief to come across something as simple, clean and focused as Pivot by the Sweden-based company Fors. Pivot brings Scandinavian minimalism to classic FM synthesis (or, to be pedantic, phase modulation), with a clean design aesthetic, all simple geometric forms and lower-case labels. Its resizable window flips between a panel for synthesis and one for modulation. The plug-in is compatible with macOS, Windows and Linux, and supports the modern CLAP standard as well as VST3 and AU.

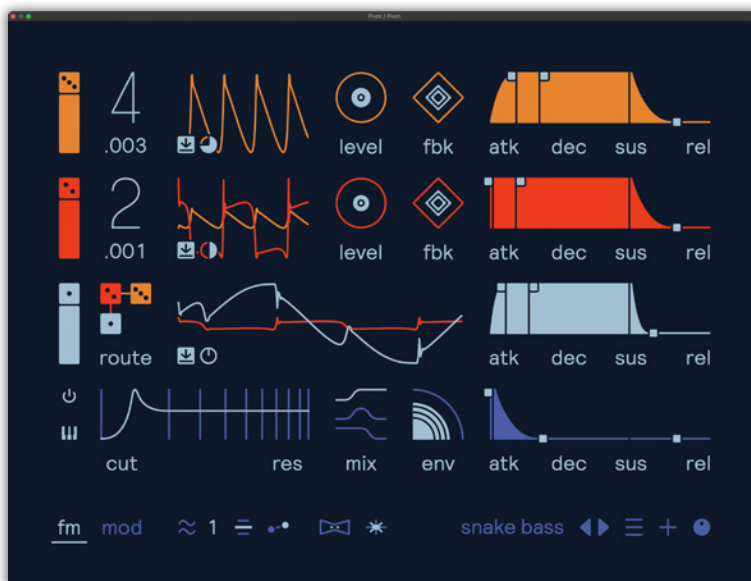
Pivot Point

The style of FM on offer here is pretty simple: three operators (one carrier plus two modulators) and two algorithms. In fact, Fors are emphatic that Pivot is algorithm-free, and that's true in the sense that you can smoothly morph between two operator layouts, so algorithm selection is not either/or. There's also a multi-mode filter (comb, acid or SVF, the latter being able to shift continuously between low-pass and high-pass) and a smattering of simple effects.

The two extremes of the operator configuration value (called 'route') are a 3-op stack, and a two-modulators-into-one-carrier tree. The route value 'pans' the third operator's output between the first and the second. This seems to cover all the important tonal ground, although you're limited to a single carrier throughout. An iconic three-dice logo shows the routing mix. Operators 2 and 3 have the usual pitch controls (course and fine), although the carrier has none, so there's no overall transpose. All three operators have their own ADSR envelope, as does the filter, and operators 2 and 3 have a level setting and feedback loop (to themselves only). The interface is pretty much devoid of knobs and sliders: you can drag control points to shape the envelopes, but the level, feedback and filter envelope follow controls look more like road signs than synthesizer knobs, although dragging up or down on them works as expected, with visual feedback.

The one concession to rich visual display is waveforms, showing the signals that are flowing between the operators and also the final pre-effects audio output. I hooked up a software oscilloscope to the plug-in and can confirm, should you be concerned, that this display is accurate.

Pivot's effects section is pretty minimal: there's a very gentle chorus for providing some stereo width ('width' being its only parameter), an effect like a bit-crusher, and a distortion effect, which I found a little underwhelming. Pivot's raw sound is pretty dry, so you'll have to rely on your DAW's suite of effects to add warmth and richness. There



Fors Pivot Software Synthesizer

Interface design has always been a barrier to FM synthesis. Can Fors' elegant approach change that?

is a unison mode which runs three voices for each note, and they can be detuned as far apart as two octaves, if you want to layer up and fill up the frequency spectrum a bit.

Flip to Pivot's modulation panel and you'll find a pair of LFOs, with a selection of shapes, and a continuous skew parameter to reshape or provide PWM. Rates can be Hertz or tempo-time, and output can be uni- or bipolar. The LFOs are per-voice, can reset on key-down or be beat-sync'ed, but there's no phase control. There are two additional ADSR envelopes available as modulation sources as well.

You can modulate from up to five internal sources (LFOs, envelopes, a note-on random generator) and five external MIDI ones (pitch, velocity, mod wheel, pressure, MPE front-back slide). MPE left-right glide tracks pitch as expected (it's not available as a modulation source), although old-school pitch-bend is

hard-wired to a two-semitone range.

The full modulation limits for pitch are pretty extreme at ± 96 semitones, which is pretty much Nyquist to DC, so you'll need to keep that in check.

May The Fors Be With You

If you're used to more sophisticated instruments, Pivot might seem a bit weak — a dry sound, relatively simple FM implementation — but on a recent project I found myself using it a lot. It's clear both conceptually and visually, does most of the harmonically rich and animated FM that you could want, and is very approachable. My review copy came with very few presets, but I don't think I used any of them, instead just grabbing the on-screen controls to make sounds from scratch each time. In a world of ultraprocessed software instruments, it's good to come across something this pure and refreshing. **///**

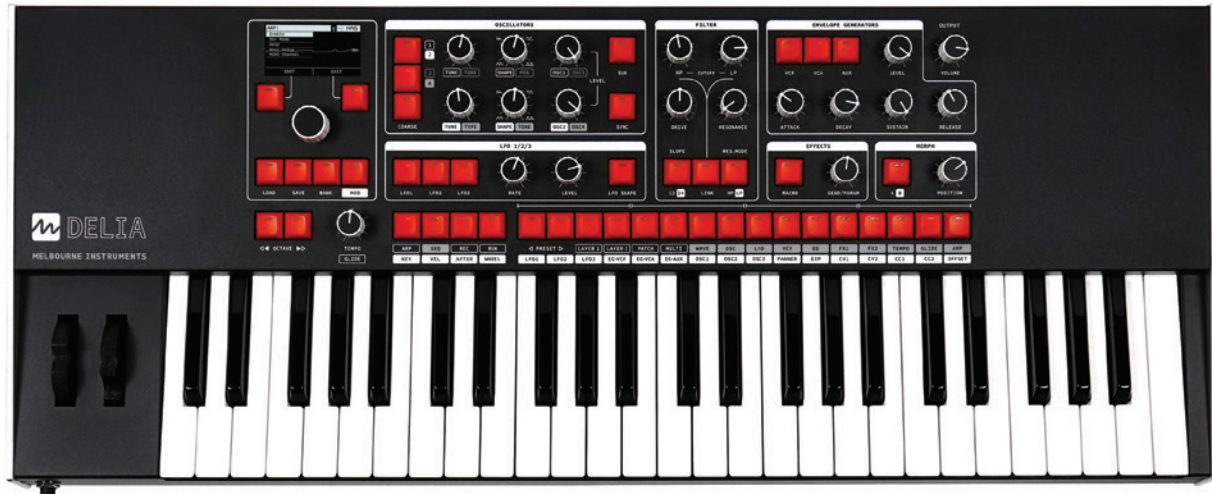
summary

Pivot is an elegant and approachable 3-op FM synth with morphable algorithm and filter, dedicated modulation matrix and MPE support in a sleek designer package.

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JOHN WALDEN

Those with a passion for media composition and cinematic music styles will be familiar with Stefano Maccarelli's various collaborations with Zero-G including the excellent *Ethera* titles. However, a further collaboration was *Elements Kepler* and this provided a Kontakt-based 'hybrid scoring toolkit' inspired by scores to films such as *Dune*, *Tenet* and *Oppenheimer*. With an impressive array of presets, a straightforward UI, plenty of sound-design options and a very accessible price, *Kepler* was very well received. Indeed, so well received that the same team are now back with *Elements Kepler 2*. So, what does this update add to *Kepler*'s sonic world?

New Tricks

As before, the full version of Kontakt (8.6 or later) is required (Kontakt Player is not supported). *Kepler 2* adds 17GB of brand-new material to the original's already hefty 33GB, expanding the palette considerably. As far as I can tell, the expanded collection of presets (now numbering over 1000) within *Kepler 2* don't duplicate those within the original release; if you have projects based upon *Kepler 1* you might — initially at least — want to keep both versions installed.

As with the first version, sounds can be built from up to three sound layers. However, *Kepler 2* delivers some UI changes to streamline access to the various sound-design options within the front ends of both the Core Designer and Texture Designer instruments. It also adds a third flavour of the engine called the Synth Designer. All three share the core features but, as explained clearly in *Kepler 2*'s manual, there are features within each engine specific to their different purposes. An expanded set of tabs is now accessible directly from the top of the UI, giving you access to various effects options (Space, Dyn and EQ, for example), envelope controls (Shape), sound layer blending (XYtab), step-based parameter modulation (Macro 1 and Macro 2), sample slicing (Slice), arpeggiation (Arp) and an option to automatically quantise any MIDI note input to a specific grid resolution (In Q). Many of these options can be adjusted on a per-layer basis and editing of each layer also offers an 'expert' mode, with further sound-design options, accessed via the square button beneath each layer's meter section towards the top of the UI.



Zero-G Elements Kepler 2

Software Instrument

Zero-G's popular hybrid scoring toolkit gets a significant upgrade.

You could immerse yourself in the extensive preset collection for a considerable amount of time. *Kepler 2* exists in the same sound world as the original, which is to say there is a huge range of textures, drones, pads, cinematic percussion, sound effects (such as risers, whooshes, impacts and braams), playable instruments including synths and basses, sounds based upon various organic sources, and a collection of vocal-based presets that offer choral textures, tribal chants, screams and throat singing, amongst others. Given the flexibility of the various engines, providing you are prepared to dig in, there is also endless scope to design your own sounds.

A significant element within *Kepler 2*'s new sample content is a collection of string instruments that utilise something Zero-G term 'Bow In Motion'. This is a rather neat legato feature where various string instruments have been sampled with long continuous performances that include audible changes in bowing direction. These have been recorded at various original tempos, but this bowing motion can also be tempo-sync'ed to your DAW. The sounds themselves are excellent and full of character but, for slow, sustained note

performances, the fact that the bowing matches the tempo of the project can add an interesting additional element of realism.

Conclusion

Kepler 2 is a very worthwhile expansion of the original concept, delivering new sounds and new sound-design options. However, the underlying concept remains a key part of what makes this library so attractive; it's a complete toolkit for media scoring within a single library. The sounds are sonically impressive, but you also have options for every element of the scoring process, from the most delicate of percussion to the darkest of drones. If you are looking for a one-stop shop for modern hybrid cinematic scoring, then hats off to Stefano Maccarelli and Zero-G: *Elements Kepler 2* delivers just that and does so at a very accessible price. **///**

summary

Elements Kepler 2 improves on the already impressive original. A great single-library toolkit for modern hybrid scoring.

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Hifiman HE600 & more

Planar Magnetic Headphones

SAM INGLIS

Hifiman sounds like one of Marvel's less well known superheroes, but they are actually a headphone manufacturer with a huge portfolio of products. The fast pace of innovation and rapid turnover of models reflects the progress being made by founder Dr Fang Bian, who is constantly striving to improve his designs. Although they make other types of headphone, Hifiman specialise in planar magnetic models, and have claimed numerous breakthroughs in driver technology.

It can be a little hard to keep up with the development of the Hifiman product range, which usually includes numerous B-stock offers, limited editions and variations on a theme. At the time of writing, the collection available in the UK comprises no fewer than 60 different headphones, including all four of the models that were sent for this review. What these all have in common is very aggressive pricing. Historically, headphones that use alternative technologies such as planar magnetic drivers have tended to occupy premium price brackets, but the Edition

XV, Audivina LE, Ananda Unveiled and HE600 are all relatively affordable. The first two retail at around the \$400 mark, with the HE600 coming in at about double that, and the Ananda Unveiled somewhere in the middle.

In The Frame

When I first tested Hifiman headphones in 2022, there was often very little in common between different models. Since then, Dr Bian seems to have been working to standardise some aspects of his designs, and the four models under review share quite a few construction features. The frames appear identical in each case, with a perforated fabric strap sitting inside a separate structural headband and cushioning the load on top of the head. The yokes from which the earcups are suspended are likewise very similar in all four models. They

permit the cups to rotate freely up and down, and a little fore and aft. In practice, this means it's relatively easy to adjust any of the four headphones to fit your personal head shape.

A final point of similarity between all of the review models is that the signal from your headphone amp enters through separate mini-jack sockets at the base of each earcup. In all four cases, alas, it's needlessly difficult to identify which earcup and which side of the Y-shaped cable is left and which is right. If I were planning to pack them away on a regular basis, I'd be making coloured labels, especially for the cables. And talking of packing these headphones away, all four of these models arrived at *SOS* Towers in smart and practical semi-rigid cases, which I haven't seen on previous Hifiman models. The Audivina LEs also come with multiple cables, including a so-called 'balanced' cable terminating in a four-pin XLR.

Once you get to the earcups themselves, clear differences are apparent between the four models. Those of the HE600 are circular, whilst the Edition XV, Audivina LE and Ananda Unveiled all employ the elongated, asymmetrical egg-shaped geometry used on the HE1000 Unveiled and Arya Unveiled models I reviewed earlier this year. The 'Unveiled' tag refers to the complete omission of any grille or louver on the outside of the earcup, a design philosophy intended to prevent sound from the rear of the diaphragm being reflected back into the earcup. Hence the Ananda Unveiled's magnet structure is

»



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» clearly visible, and they also ship with cloth bags and magnetic shields that can be placed over the earcups to protect them from small particles. By contrast, the Edition XV and HE600 both look more or less conventional from the outside, with the magnet structure covered by an attractive hexagonal mesh. The Audivina LEs are very much the odd ones out, being a closed-back model. Their rigid black plastic case gives them a very '70s appearance.

Theme & Variations

All four models under review are passive, planar magnetic designs with relatively similar specifications. Nominal impedance is between 12Ω and 30Ω, with sensitivity measuring between 92 and 96 dB. The stated frequency response is, as ever, vast, but no tolerances are given. In practice, they won't be as loud as a typical pair of moving-coil headphones for a given signal level, but they're perfectly capable of being driven well by a fairly mundane headphone amp.

I've heard a lot of very impressive open-backed planar magnetic headphones, but the intrinsically dipolar nature of this type of driver makes it tricky to develop a good-sounding closed-back planar design. When the driver is pushing the same amount of air both ways, something has to be done

with the sound emerging from the 'wrong' side; and if it can't be allowed to escape into free air, some other means needs to be found of stopping it bouncing around and interfering with the wanted sound from the 'right' side of the driver. I'm not sure what approach Hifiman have taken with the Audivina LE, but the outer cases are about an inch deep, and presumably house some sort of damping material.

One of Dr Fang Bian's long-term design goals has been to improve the treble response of planar magnetic drivers, which historically were rather lacking in high-frequency extension. He reached the broad goal of designing a bright-sounding planar headphone

“For me, the crown goes to the HE600s, which sound both more balanced and smoother than any of the other three models.”

some years ago, but in some of the older open-backed Hifiman models, this brightness could be a little uneven or peaky. That same phenomenon is apparent in the closed-back Audivina LEs, which to my ears exhibit a sharpness somewhere in the 5kHz region. Shakers, hi-hats, vocal sibilants and the like can be uncomfortably prominent, with a slight 'whistling' quality apparent in some case.

This peak sits on top of a basic tonality that is already on the bright side of neutral, and to my ears, the bass and low midrange seemed a little subdued. Having said that, the Audivina LEs do manifest many of the good qualities that we associate with planar designs: they're clean, with a punchy dynamic response that puts across transients in a very lifelike way, and they present a broad and coherent stereo picture. I probably wouldn't want to mix on them without corrective EQ from something like Sonarworks' SoundID, but I can imagine them giving good service once the frequency balance has been adjusted.

Clean Air

What, then, of the three open-backed models? Unsurprisingly, there's a fair amount in common between them, including all of the generic strengths of the planar design mentioned above. But they certainly don't sound the same. The Ananda Unveiled struck me as the brightest of the three, with a sound

signature that somewhat emphasises the treble and upper midrange over what might be called the 'true' midrange. In that respect, they're not so dissimilar to the Audivina LE, except that the presence lift seems a lot smoother in this case. In an ideal world I'd probably still want to use them with something like SoundID or RealPhones, but I think you could fairly easily learn to accommodate the ways in which they diverge from neutrality. At the other end, the bass is quite lean and clear, but perhaps not as deep as you'd hope for from a planar magnetic headphone.

The Edition XV has somewhat more going on in the 100-200 Hz

department, and an overall timbral focus that seems slightly lower in the frequency spectrum, albeit still favouring the upper midrange.

On some material this

struck me as a little uneven, and it sometimes exaggerated the 'knock' of a bass drum or a honky resonance in a guitar. However, this is a relatively subtle phenomenon, and given that the Edition XV is the most affordable of the four models on test, I think they represent excellent value for money.

For me, though, the crown goes to the HE600s, which sound both more balanced and smoother than any of the other three models. Yes, they're more expensive, but by the standards of high-end planar designs, they're not particularly costly in an absolute sense. And whereas I'd have some qualms about mixing on the other models without software correction, the HE600s would have my full confidence. They're detailed but not over-bright. They are warm without unduly exaggerating the low midrange. And they have all the aforementioned positives that are common to other Hifiman models, and to planar magnetic headphones in general. Hifiman may not have a cape or a mask, but they do have the superpower of building high-quality headphones at extremely competitive prices! **///**

summary

Hifiman are committed to delivering a lot of headphone for your money, and the HE600s in particular represent excellent value.

\$ HE600 \$799; Ananda Unveiled \$549; Audivina LE & Edition XV \$399.

W www.hifiman.com



Dear Santa

This year, I've been:

Mostly good

Exceptionally creative

Full of great ideas



Ooops! Coffee spill!!

But I haven't been recording because I still don't have a Scarlett



So, please, if you can bring me:

★ Focusrite Scarlett 2i2 – So I can finally record crystal-clear vocals and work on winning a Grammy from my bedroom.

★ Novation Launchkey MK4 – Because my ideas deserve velocity-sensitive keys, not a \$20 keyboard from 2009.

★ ADAM Audio D3Vs – I need to feel the bass and hear the truth. Also, they'll look cool on my desk!

★ ADAM Audio H200 Headphones – For when I want studio detail, but my roommates want silence.

I don't need socks...
I need incredible sound quality!!



Sincerely,
An Aspiring Audio Legend

Focusrite®



novation



ADAM AUDIO

Active DI Interface

Strymon PCH

This pedalboard-friendly DI interface has been designed specifically for interfacing amp modellers and electronic instruments with PA and recording systems.

DAVE LOCKWOOD

Even in today's ever more high-tech live-sound world, a guitar player arriving at any gig with a 'provided PA' is most likely to find that the engineer's default option will be a single line to front-of-house and an SM57 to place in front of the speaker. All well and good if you've turned up with a conventional guitar

amp — you'll be a mono source arriving at the console at the expected 'mic level'. Not so good, however, if you've brought a digital amp modeller and pedalboard loaded with your lovingly crafted stereo effects. What usually happens then is that the 57 gets somewhat reluctantly swapped for a single-channel DI box and the engineer departs without answering your questions about what kind of level you should be sending...

Amp modellers are usually capable of outputting a very wide range of levels, from a hot 'pro' line level down to mic- or passive instrument level, and may have anything from balanced XLR or TRS sockets to unbalanced TS jacks. You might imagine that you'd always be pretty safe with a nice, strong line-level signal on balanced XLRs that you can connect directly to any stagebox, but there's still plenty that can go wrong in that scenario. The chances are that the stagebox will connect to the mic inputs on the console, with line-level signals expected to be handled either with a separate TRS input, or maybe just a pad switch to drop the sensitivity of the mic inputs. It's also sadly true that not all XLR-equipped modellers can deal with being sent phantom power,

and not all PA consoles have phantom power switchable per channel.

On the whole, you are often better off just playing safe and using a DI box that will drop your modeller's signal down to mic level... except not all DI boxes are created equal! You could be clipping the DI box before your signal even gets to the console, and if you've taken a split to a local monitor from before the DI, you won't even know that your FOH sound is compromised.

Who Needs A 'Specialised' DI box?

The widespread and ever-increasing use of virtual amps in live sound and recording over the last few years has led to the development of a few dedicated 'pedalboard amp' interfaces, notably Pinstripe Pedals' DISO and Walrus Audio's Canvas, and they are joined now by the PCH from premium pedal-makers Strymon. About the size of a standard Boss pedal, the PCH is an active, high-headroom, stereo DI interface with a pair of balanced, low-impedance XLR outputs, a pedalboard-PSU-friendly 9V DC input (internally converted to 24V), and, unusually, an onboard headphone amp. I say 'pedalboard-PSU-friendly' because it's 9V, but with the one caveat that it requires a fairly hefty 500mA current supply. »

Strymon PCH

\$279

PROS

- Excellent signal integrity.
- Flexible routing and connections.
- Great-sounding headphone amp.
- Bullet-proof build quality.

CONS

- An aux input would have been nice.

SUMMARY

Strymon's PCH is more than 'just another DI box'. When you've spent serious time and money crafting a great-sounding modelling rig that you hope will be feeding into a great-sounding PA system or recording rig, the 'bit in the middle' really matters: a signal chain, like any other chain, is only as strong as its weakest link!

_KeyStep mk2

Control anything.
Sequence everything.

The musical nerve-centre for hybrid setups. Spark ideas in your DAW, pilot any synth or patch modular without breaking your flow.

Capture a riff, turn it into a sequence, then mutate it. Generative tools and real-time control work in synergy to keep ideas evolving.

Where will KeyStep mk2 take you?



ARTURIA

_The sound explorers

» Left and right inputs are high-impedance, unbalanced, quarter-inch jacks, described by Strymon as 'virtually unclippable', and indeed proving to be so in testing — these inputs will handle full line level from a modeller with further gain from downstream effects as well, although I wouldn't advise running a rig that way! A second pair of unbalanced jacks provide a buffered 'thru' output, which might be used to feed the input signal to any on-stage amplification as well.

Connecting to just one of the inputs sends the same signal to both thru outputs, and both sides of the headphones, too. If you are going into the PCH in stereo with a connection to both sides, you can sum the thru outputs to mono while the main outputs remain stereo. If you have a modeller that can output a speaker-simulated signal on one channel and a 'no-cab' signal on another, you can connect to the PCH in double-mono, sending the speaker-sim channel to the PA via the appropriate XLR output, whilst picking off the 'dry' signal for on-stage amplification with a conventional guitar cabinet by using the matching thru output.

The XLRs benefit from both a ground-lift switch, for combating potential earth-loop hum, and mic/line level switching, giving you back control over optimising your front-of-house feed without having to change your local monitor feed levels, which may be set just right for your personal monitoring already.

At \$279, the PCH is not a trivial purchase, so why not just get a stereo DI box that you have verified as having enough headroom and attach that to your pedalboard instead? Most passive boxes will give you earth-lift isolation and thru outputs, and active boxes may give you input level matching as well, but the latter will need powering, which means you are at the mercy of the console providing phantom, or using batteries. What you may not have, however, is any summing facility, and you certainly won't have a very



■ Not many DI boxes include a headphone output! This allows you to audition the signal after all your pedals, before it goes to the FOH mixer.

nice headphone amp included. Being able to listen to the signal right at the end of your pedal chain gives you certainty as to what you are sending to the PA and is also useful for silent practice using your modeller and subsequent effects. Even if your modeller itself has its own headphone output, you won't be able to hear pedals patched in further downstream. The absence of an aux input on the PCH for some kind of playback device to play along with will be a disappointment to some — although you can sort of get around that by using one input for your guitar signal in mono and the other for a playback source. It sounds odd for about five seconds and then you simply get used to it. But if playing along to stereo tracks with your full rig in stereo is a big part of your practice routine, it's probably worth getting a small mixer and doing the job properly.

Just One Thing...

I like units that are specified and configured to do a specific job really well, and I think Strymon's PCH fits that

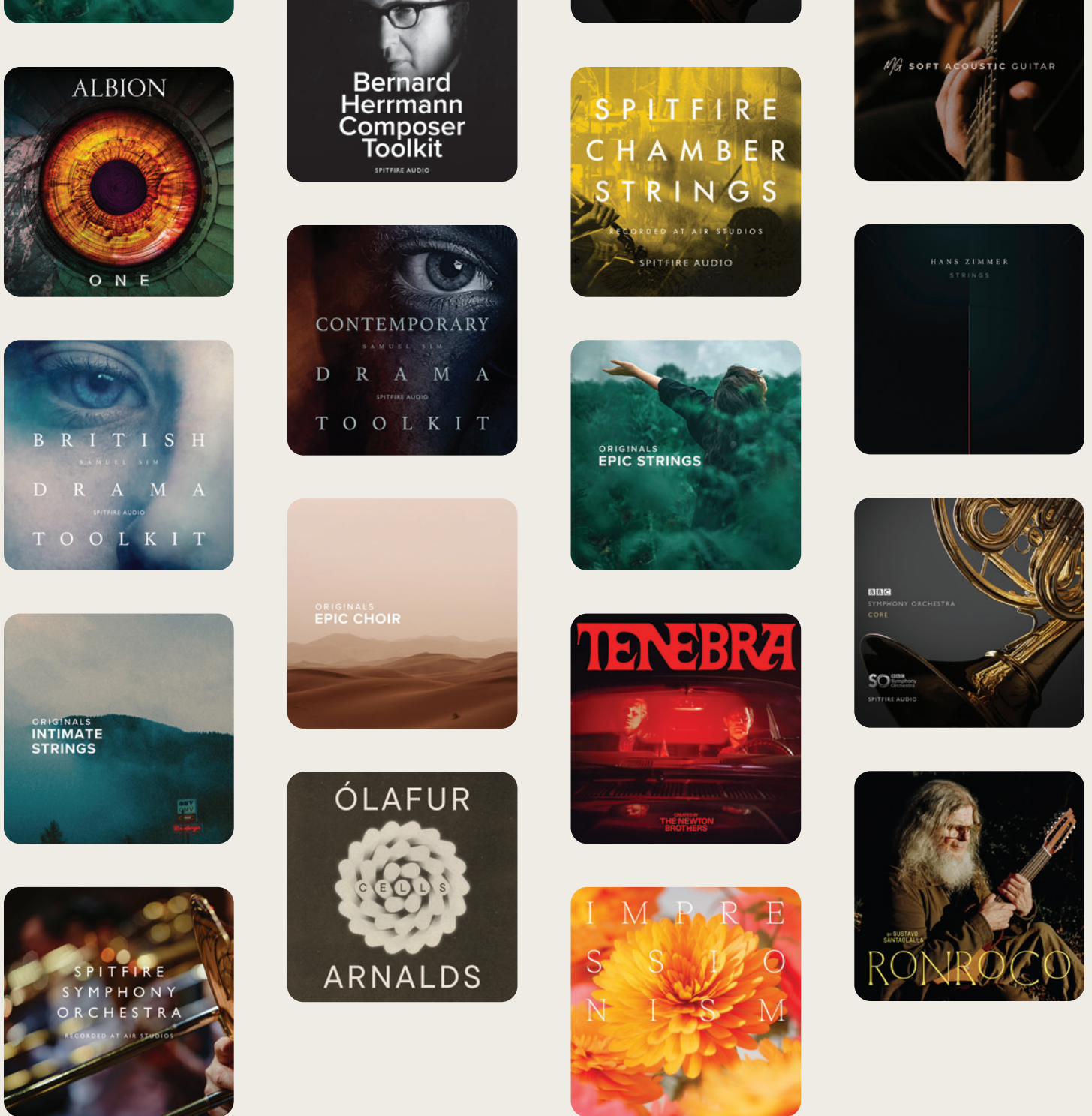
brief perfectly. With a rugged, compact shell (with a nice flat base for securely attaching to a board) and high-quality connectors and components, it out-specs most of what will go into it or receive its output. If I could change just one thing about it, it would be to add a summing facility to the main outputs. Even when FOH can spare you a second channel, I am dubious about the value of running primary signals (as opposed to effects) in stereo into a PA where a large part of the audience will be hearing mostly one side or the other. Stereo on stage, however, with a pair of full-range monitors enveloping the player with a supportive ambience, is a performance-enhancing delight. And stereo is almost an 'essential' for optimising in-ear monitoring, I find. With the PCH, I can 'mono' the local thru feed that I usually want to be stereo, but can't 'un-stereo' the PA feed that I want to be mono. The designers at Strymon know what they are doing, however, and I have no doubt that the PCH will hit the mark for the vast majority of potential users, so maybe that's just an issue for me.

Overall, this is a great solution to have available for users of all electronic sources, from virtual amps to drum machines and keyboard rigs, and I guess you could always use it 'backwards', sending the stereo XLRs to stage monitors and using one of the mono-summed thru outputs for the PA! ■■■



■ The main XLR outputs can be set to deliver a mic or line signal.

\$ \$279.
T Strymon +1 805 468 8788.
E info@strymon.net
W www.strymon.net



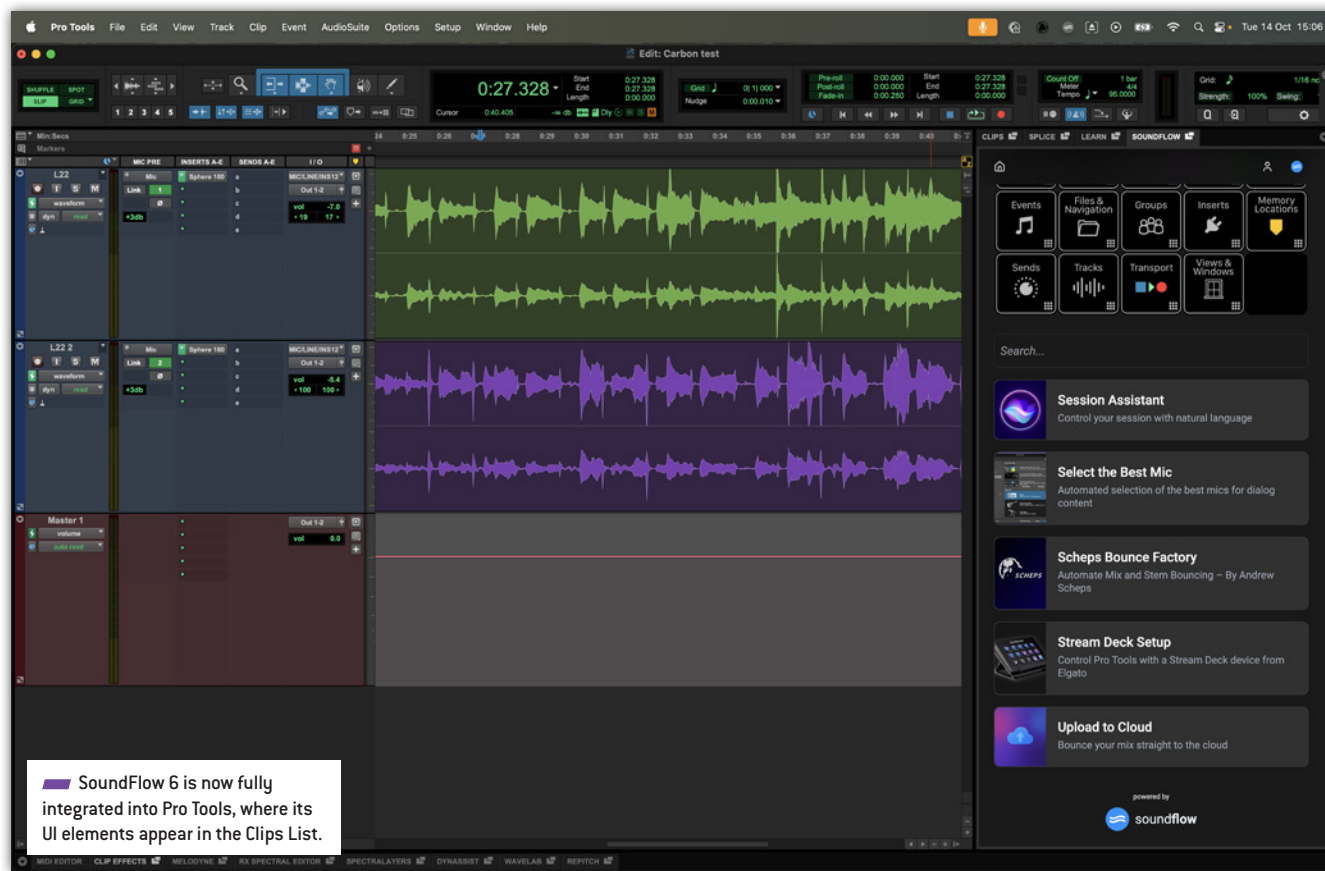
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SAVE UP TO 50% on our award-winning libraries, plus unlock an extra 20% saving when you join our newsletter.



Sign up to
our newsletter

SPITFIRE AUDIO



A ground-up reinvention turns SoundFlow from an advanced macro engine into an intelligent, context-aware virtual assistant.

SoundFlow 6

DAW Control Software

JULIAN RODGERS

Since its introduction in 2017, SoundFlow has grown from a clever macro utility for Pro Tools into a fully featured automation environment. Its user base includes many working professionals who value its flexibility, problem-solving power, and the active online community that supports users through issues, answers questions and shares workflows.

Originally developed by engineer Christian Scheuer to automate repetitive tasks in Pro Tools, SoundFlow exploited the Accessibility features in macOS to trigger custom commands and build macros. Its roots in real-world editing and mixing workflows explain why it has always been strongly aligned with Pro Tools. But despite that close association, it was never limited to that DAW: SoundFlow also offers official support for other applications such

as Ableton Live, Logic Pro, iZotope RX, Sibelius and Media Composer, which can benefit from pre-made commands and 'decks' in the same way as Pro Tools.

SoundFlow can be used to create automation for any software on the Mac, and there is an active community of users creating and sharing their custom solutions online. However, if a piece of software has not implemented macOS Accessibility features, then automation is limited. It can also be used for more mundane desktop tasks — for example, launching web pages or switching apps at the touch of a button.

This breadth of potential has long been one of SoundFlow's greatest strengths. Its openness comes at the cost of initial complexity, but also opens the door to almost unlimited customisation. Many users build bespoke control decks, shortcuts and workflows to automate nearly every aspect of their daily computer work — from session setup to deep editing, from

batch processing to web launcher decks of their most-used pages and tools.

Version 6 marks a clear step change. It builds on that deep base of user-driven automation and the tight partnership with Pro Tools, and then reimagines how automation can support creative work. With a new engine, new ways to build solutions, and the use of AI, SoundFlow 6 feels not just like a more powerful tool, but a more capable assistant: one that understands context, adapts to session needs, and helps users get more done with less friction.

User Perspective

I've been using SoundFlow since version 5, which also represented a significant deepening of capability compared to earlier versions. Whereas prior editions were useful primarily for repetitive keystroke automation, SoundFlow 5 introduced pre-built scripts, macros and decks that

substantially changed how you could work in Pro Tools. As part of this, the SoundFlow team collaborated with Avid to deliver a curated package of pre-built macros to Pro Tools users, effectively offering more sophisticated alternatives to standard Pro Tools keyboard shortcuts.

However, the real instant impact in workflow comes from SoundFlow's decks: grids of custom buttons that you can trigger from a touchscreen device such as an iPad or, more effectively, from a hardware controller such as a Stream Deck. Even without creating your own custom content, you can immediately access powerful commands and workflow enhancements.

The SoundFlow app's built-in usage-tracking statistics make its value very clear. After moving from a touchscreen-only setup to a dedicated Stream Deck, my use of SoundFlow increased dramatically. The tactile feedback, visual cues and context-aware displays on each Stream Deck button combine the best of hardware and software design. If you're serious about workflow efficiency, buying a Stream Deck pays dividends quickly.

Part of what makes this easy is the pre-built, bundled SoundFlow content for Pro Tools users. Unlike the earliest versions, where the emphasis was on installing your own macros, the packaged content means you can be productive immediately. You don't have to start from scratch. Users have shared their own decks and macros for years, but having pre-installed, ready-to-go content essentially eliminates the learning curve for many.

That said, for a technically minded user, SoundFlow's depth remains both enticing and frustrating: you could, if you wanted, build everything yourself, and go deep into scripting, custom commands and detailed workflows. You probably never will learn all of it, but that's not a workflow problem — it simply means that the tool has far more to give when you want it to.

SoundFlow In Pro Tools

In earlier versions, SoundFlow always operated as a parallel layer alongside Pro Tools. With version 6, for the first time, SoundFlow is embedded within Pro Tools itself. Controls, decks and commands are located where they belong: front and centre, in your DAW, and immediately accessible to every user.

In terms of user uptake, one of the most significant changes in version 6 is that you no longer need to install SoundFlow separately. It is now installed by default with Pro Tools 2025.10. The onboarding process has been stripped of friction: there are no extra downloads, no external installers, no separate login screens.

For new users, account creation and login happen directly from inside Pro Tools, using either a dedicated SoundFlow account or your existing Avid credentials. Immediately, you get access to pre-installed decks (grids of buttons to access commands), as well as very instructive built-in tutorials. For existing SoundFlow users, the transition is easier: there's no longer a need to manually install the app.

SoundFlow 6

PROS

- Deep integration in Pro Tools, with no separate install or setup.
- New SFX engine enables powerful features and Windows support.
- AI-driven Flows enable adaptive, context-aware automation and chat-based interfacing.
- Interactive tutorials hugely improve onboarding and learning.
- Retains compatibility with existing SoundFlow 5 scripts and workflows.

CONS

- Learning curve remains steep for advanced customisation.
- SoundFlow Editor currently unavailable on Windows at launch.
- Fully realising its potential requires SFX implementation by third parties.

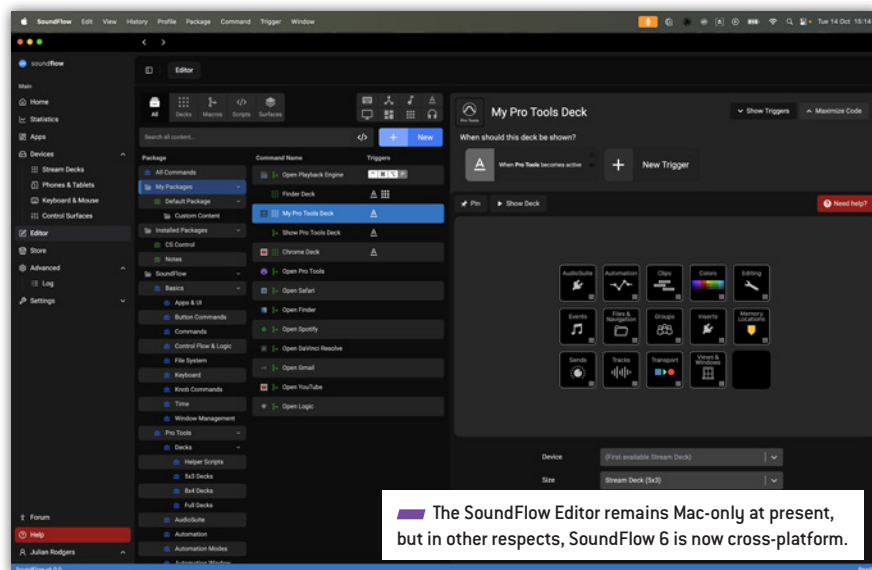
SUMMARY

SoundFlow 6 marks a major step forward for workflow automation in Pro Tools and beyond. The new SFX engine, cross-platform capability, and AI-driven Flows take SoundFlow's possibilities far beyond previous versions. What it can do today is impressive, but the potential it opens up for future development is huge.

Inside Pro Tools, SoundFlow is accessed from a tab located in the Clips List sidebar. From here you can launch decks of commands or macros, or detach them into floating windows which can be placed on another screen if desired; use the search bar to find commands, macros or decks by name; and discover and run the new

»





» non-deterministic sequences called Flows, of which more later.

Triggering SoundFlow actions isn't limited to clicking buttons in decks: you've always been able to assign keyboard shortcuts, MIDI events, OSC messages or even EuCon input. SoundFlow intercepts and handles key presses before they reach Pro Tools, so takes priority over Pro Tools' own keyboard shortcuts. Because Pro Tools' Keyboard Shortcuts window is searchable, it's relatively easy to avoid conflicts: before assigning a shortcut in SoundFlow, check whether the key is already used, and if so, whether that use matters to you if you sacrifice it. The built-in visual keyboard map helps with this.

Another strength is that your SoundFlow settings live in the cloud. This means that on guest systems or different machines, you simply log into your SoundFlow account and your custom shortcuts, decks and Flows follow you. A corresponding down side is that SoundFlow requires an active Internet connection. For most users, especially in music or general post workflows, this isn't a major barrier. In highly restricted or offline environments, SoundFlow does offer solutions, but added complexity may be a consideration.

The Heart Of It

SoundFlow actions in version 6 look and feel faster and smoother than before. In earlier versions, they often visually replayed menu navigation and UI interaction, so you could see what was happening. In version 6, they often execute in the background or invisibly. The results are faster, more fluid and less visually distracting.

For deeper tasks like creating your own commands, writing scripts and editing decks and Surfaces, the full SoundFlow application and editor remain available. Pro Tools 2025.10 installs these components in the background along with the bundled content. From the menu bar you can launch the SoundFlow editor to undertake customisation, scripting or complex controller builds. Custom control layouts called Surfaces are available to power users, though casual users will never need to create them.

Importantly, many of the valuable benefits of SoundFlow can now be accessed without ever opening the editor. This significantly reduces the barrier to entry. Expert users still get full control, but non-technical users can start right away and reap real productivity gains.

In short, SoundFlow 6 is no longer a bolt-on adjunct. It feels native to Pro Tools.

It is accessible from the moment you open a session. It integrates deeply, adapts to your workflow, and offers both simplicity for newcomers and depth for power users.

SFX

In previous versions of SoundFlow, mouse clicks, menu navigation, key presses and UI control were all triggered via operating system Accessibility hooks. That meant that SoundFlow was inherently Mac-only. Version 6 is built instead on a new underlying engine called SFX. SFX is a framework developed by SoundFlow and it replaces, though does not eliminate, the older Accessibility-based approach. Existing macros remain compatible, so long-time users needn't fear breaking their workflows. But SFX offers a fundamentally different and deeper way of interacting with applications.

When I first heard about SFX, I assumed it was built on top of Avid's SDK for Pro Tools, but it isn't. It uses a different communication method that can be used to automate any supported application. The result is both higher performance and greater fluidity. Some actions execute without noticeable UI animation, and tasks complete more cleanly. And because SFX does not depend on macOS Accessibility, Windows users are no longer excluded, although at the time of writing, the full SoundFlow editor for creating custom scripts, decks and Surfaces is only available on macOS. Windows support is expected to follow.

In practice, SFX unlocks the potential for automation that is non-deterministic, content-aware and context-sensitive. In earlier versions, you fired a macro, and it ran the same sequence of steps every time. With SFX, automation can react

»

Pricing & Tiers

SoundFlow 6 is available in four distinct tiers, scaling from casual use to full professional integration. For Pro Tools 2025.10 users, SoundFlow is included at no additional cost courtesy of the new Free tier. This pre-installed version provides access to bundled decks, tutorials and basic automation features. While it omits some of the more advanced capabilities of the paid plans, such as Stream Deck integration, it remains genuinely useful, with over 1700 commands available from the SoundFlow panel and the ability to assign custom keyboard and MIDI triggers to those commands in the SoundFlow Editor.

The first paid-for tier is SoundFlow AI, priced at \$10 per month or \$100 per year,

which introduces the AI-driven features and Flows. Like all cloud-based AI services, these functions carry real computational and energy costs, so it feels reasonable that access is subscription-based. SoundFlow Pro, at \$12 per month or \$120 per year, corresponds to the full-featured edition familiar to long-time users, including scripting, macro creation, Surfaces and integrations with Stream Decks, phones and tablets. Finally, SoundFlow Pro AI, the flagship plan at \$20 per month or \$200 per year, combines every feature from both Pro and AI tiers.

Existing subscribers retain their current pricing and feature sets, ensuring continuity for legacy users.



moog



A MACHINE FOR
DEEPER MEANING
* * * * *
* * * * *

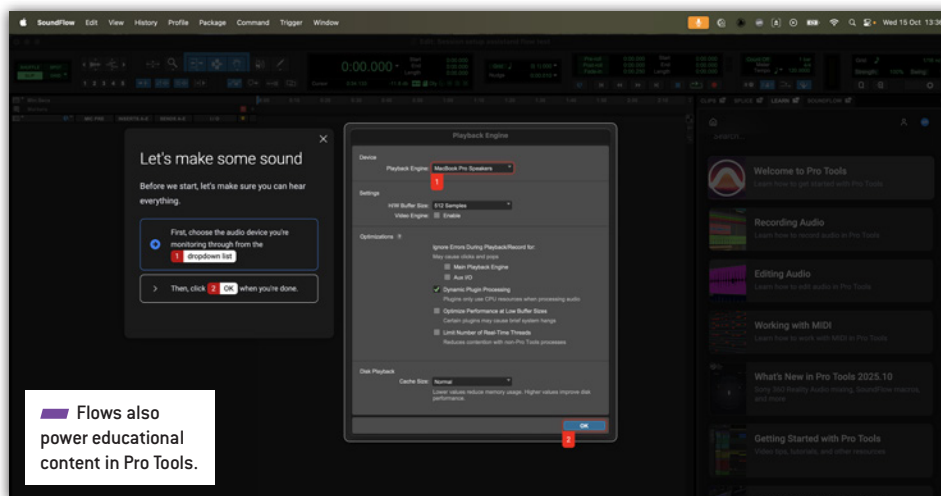


Tutorials & Learning

As described in the main text, the Session Assistant Flow can be instructive for new users because it doesn't just perform tasks, it shows you how they're done. That concept has also been developed in SoundFlow 6 and Pro Tools 2025.10 to deliver an entirely new style of tutorial system. Traditional linear video walkthroughs remain part of the package, and in some cases, a one-way video demonstration is exactly what's needed. But video training can only ever show one example, at one pace, with one outcome.

The new SoundFlow-powered tutorials take a completely different approach. They are interactive and built on SFX-based Flows, meaning they are dynamic, adaptive and capable of responding to what you do inside your own active Pro Tools session. Rather than watching a video of someone else using Pro Tools, you can learn in Pro Tools, with the software itself guiding you through each step.

From the Learn tab in the Clips List sidebar, you can access a collection of both linear video and interactive Flow-based tutorials. Examples of the latter include a Getting Started tutorial and Flows on recording and editing audio and working with MIDI. Because these tutorials are driven by the same



Flow technology that underpins automation in SoundFlow 6, they can interact directly with the interface. They can highlight the correct menu, prompt you to click or select an item, wait for your action before proceeding, and even adjust the next instruction depending on what you actually do. If you make a different choice, the tutorial adapts accordingly.

As someone who has worked as a software trainer, I find this development remarkable. One of the core challenges in teaching any complex application is that every

learner progresses at a different pace. Video lessons move at a fixed speed. Manuals assume prior understanding. But an adaptive tutorial proceeds in step with the learner. It teaches by doing, not by showing: a principle that any effective teacher recognises as crucial to genuine understanding.

These interactive tutorials are already being used to support onboarding for new Pro Tools users. They guide you through basic setup: how to get sound from your interface, how to route tracks, how to record, and how to navigate the

Pro Tools environment. For me, this is one of the most exciting developments in SoundFlow 6. Flows are impressive as automation tools, but their application in teaching represents something genuinely new. This is a training system that behaves like a knowledgeable assistant, embedded within the DAW, responding to your actions and guiding your progress. For newcomers, it could transform the learning process. For experienced users, it demonstrates just how flexible the underlying SFX and Flow technologies really are.

» to what's actually in your session: it can make decisions, adapt, or ask you for clarification before proceeding.

Because of this, I believe SFX is not merely a technical upgrade, but a new foundation for how automation can operate in professional audio. The potential is there for SFX, rather than any single deck, command or feature, to be the long-term legacy of SoundFlow 6.

SFX can work with any application, but needs to be implemented at the code level by the developer. Pro Tools is the first to integrate SFX into its codebase, but a significant number of other companies have expressed an interest, and the first plug-in integrations are expected early next year, with more to follow. A turnkey integration for some of the more common frameworks used by plug-in developers already exists, making plug-in support more easily achievable for developers who use these frameworks, and with Pro Tools, Live and particularly

Logic all on board with SFX, it's realistic to speculate that DAWs such as Cubase, Studio One and Reaper might follow soon.

SoundFlow AI & Flows

The new SFX engine in SoundFlow 6 provides speed, stability and cross-platform compatibility. But what's the headline new feature? The one that changes what SoundFlow actually is? That has to be Flows.

As already outlined, SoundFlow 6 moves beyond deterministic automation — the fixed, step-by-step execution of scripts and macros — and introduces non-deterministic, context-aware automation sequences capable of making intelligent decisions. In other words, Flows can respond to the situation they're presented with, rather than simply replaying the same routine every time.

Unlike macros, scripts and earlier SoundFlow automation, Flows are adaptive, content-sensitive and responsive. Their behaviour changes depending on where

they're used and what material they're working with. They can even react to user actions in real time, adjusting their next steps based on what the user does or the data they encounter. SoundFlow 6 includes several example Flows that make the difference immediately clear.

For example, the Best Mic Flow analyses multiple microphone recordings of the same source and selects the best capture based on quality rather than level. Instead of treating the loudest track as the best, it uses AI to evaluate clarity and separation, assigning the cleanest takes to each mic. This goes far beyond what could be achieved with gates, expanders or basic scripting.

Upload to Cloud demonstrates how Flows can manage routine, multi-step tasks. It uses a conversational interface to guide you through exporting audio, configuring routing, and sending the result to a cloud service. Once complete, it can even import timestamp comments

»

**ONE INSIDE KICK MIC.
ENTIRE KICK DRUM CAPTURE.**



**MADE FOR
USE IN
KICK**



**KICK
MIC**

END-ADDRESS PRESSURE GRADIENT
FET CONDENSER MICROPHONE

[LAUTENAUDIO.COM/KICK-MIC](https://lautenaudio.com/kick-mic)

» from the remote service back into your Pro Tools session as markers.

Session Assistant shows the potential of conversational, AI-assisted interaction. It uses either typed or voice input to carry out session setup tasks such as creating and naming tracks and moving tracks into track folders. If it needs clarification, it asks, and as it performs actions it shows them on screen, allowing the user to learn by watching. This dual purpose, as both automation and education, is a subtle but important change. It means SoundFlow isn't just doing things for you; it's showing you how they're done.

Session Assistant has launched with a limited feature set, but this will grow over time. SoundFlow says this growth will be prioritised based on user feedback as much as possible. At present the skills it possesses are focused on track management and control, and playback control. Of course, as it uses a chat-based interface, you can just ask it what its skills are and it will list them. If you ask it to do something outside its current skills, it will offer guidance on how to do it instead.

These examples are early indications of what Flows can achieve. In use,

it's not perfect, but this is a very early implementation and experience tells us not to draw conclusions about AI too early. It gets better very quickly so I'm not going to focus on things it can't do today. Just as the SoundFlow ecosystem of commands and decks has grown organically through community sharing, new Flows will continue to appear over time. Because Flows are adaptive and content-aware, the range of possible uses is far broader than any system of fixed macros could offer.

The implications go beyond convenience. Flows represent a conceptual shift, changing SoundFlow from tool to assistant. They demonstrate how AI can

“Describing what SoundFlow does is like trying to explain what a computer was for in 1982.”

enhance creative processes without undermining user control, handling repetitive, rule-based work while leaving the creative decisions in human hands. For experienced users, that means less friction and more focus. For newcomers, it means immediate contextual help inside the DAW.

Conclusion

SoundFlow's integration within Pro Tools marks a significant shift. SoundFlow is no longer an external utility sitting alongside the DAW. It's now part of the application itself. While technically still optional, it now lives where it belongs; inside Pro Tools, at your fingertips. The significance of that can't be overstated.

The most impressive element of this release is SFX. It isn't a refinement, but a complete re-engineering of how automation happens. It's faster, and opens up possibilities that even experienced users may not yet fully grasp. SFX feels like the foundation of something much bigger: the technology on which new intelligent, adaptive automation can be built. From everything I've seen, it's both elegant and powerful.

Flows sit on top of that foundation and showcase the possibilities. While macros, scripts and decks from SoundFlow 5 were already invaluable, Flows belong to a different category entirely. They feel like the future. It's a shift from faster execution to intelligent assistance.

Equally importantly, SoundFlow 6 extends what already works, without breaking compatibility or alienating

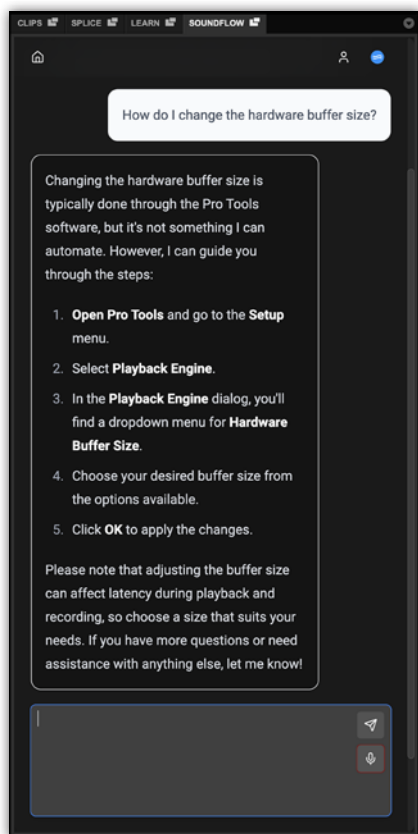
established users. Professionals depend on stability, and SoundFlow's developers clearly understand that progress should never come at the expense of backward compatibility. SFX can be invoked within existing scripts by adding a single line of code, which illustrates how carefully the transition has been managed. There's no arbitrary divide between pre-SFX and post-SFX workflows: just continuity and growth.

For new users, onboarding has never been easier. Many of us have encountered excellent technologies that never became part of our day-to-day practice simply because they required effort to adopt. By being embedded directly in Pro Tools, SoundFlow 6 sidesteps that problem. It's right there, visible and available, and once people try it, it's likely to stay in use.

Of course, no tool this flexible is easy to explain. Describing what SoundFlow does is like trying to explain what a computer was for in 1982. The only honest answer is “What do you want it to do?” That breadth can be both its strength and its frustration. It's so open-ended that even long-term users may never explore all of it — but that's not a failing. The point isn't to master every part, but to find the parts that make your own work easier, faster and more consistent.

There will always be users who resist adaptive automation, preferring to remain entirely hands-on. And that's fine. Nothing in SoundFlow 6 is compulsory. In the same way that some engineers still prefer to work on tape and consoles, there's room for every approach. But for most professionals working in modern post or music production, SoundFlow now represents a better way to interact with the tools we already use.

In final assessment, SoundFlow 6 is one of the most significant and potentially transformative developments in the way we interact with our DAWs in years. It feels like an evolution that, in hindsight, seems inevitable yet it's still remarkable in its depth and execution. I don't say that lightly. I see many new products; very few impress me, and fewer still surprise me. SoundFlow 6 has done both. Its scale and potential are enormous, and I suspect we're only beginning to glimpse what it will make possible. **///**



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Roland TR-1000

Drum Machine



Elegantly combining analogue and digital, and balancing programming with performance, the TR-1000 is the best drum machine Roland have ever made.

SIMON SHERBOURNE

Music production gear is often designed with aggressive price points in mind and that means compromises, but the brief for a true flagship product should be more of a 'give them what they want and we'll figure out the money later' vibe. Initial reactions suggest that the TR-1000 is indeed the pro drum machine we wanted from Roland. Join us as we explore Roland's vision of an analogue drum machine for modern times.

The Big Picture

The TR-1000 expands both physically and functionally on Roland's primary

general-purpose drum machine of recent years: the TR-8S. It adds analogue drum voices to the digitally synthesized and sample-based sources, new features like sample slicing and bpm matching, and a much improved user interface.

The instrument follows the layout of the TR-8S, but moves away from the black plastic Aira build to a more serious-feeling metallic silver chassis. There's some resemblance to the MkII Elektron flagships; while the step keys reference the 909 they are more like Elektron's, and there's a modulation crossfader like the Octatrack.

Comparisons with the Analog Rytm were inevitable as it's the reference point in my studio for a modern hybrid drum machine. But it's fair to say that *everyone* has been influenced by Elektron's conventions in recent years.

And if Roland had asked me what I wanted it would have been the TR-8S's form factor with the Rytm's functionality. In the end the 1000 has kept all of what makes the 8S unique, and crossed over into the territory of multiple rival units.

Sat next to an 8S or Rytm the TR-1000 is massive. It's a heavy, chunky wedge built to be a studio centrepiece and pro touring instrument. The thickness and raked panel

make room for that analogue synth circuitry as well as an internal power supply, and the resulting rear-panel space has been stocked with a full set of split track outputs, full-sized MIDI, USB ports, some analogue trigger/CV connectivity and more.

The panel design comes from the TR-8S, which itself was an evolution of the 808/909's track columns and step keys. As we've said before, the faders are a defining feature, typically slammed up and down in place of mutes rather than used for setting balances. It's hard to explain why this makes these machines so playable — it just does.

Fader count is down from 11 to 10, but notice channels 1-4 are double width with six knobs. These are dual-layer tracks; more on that in a bit. There's a proper screen instead of a text display, supported by six push encoders. This, along with the wealth of dedicated buttons and a generally better-thought-out UI make working with the TR-1000 faster and easier. You're much less reliant on assigning knobs as you can achieve most sound design and editing tasks quickly from the screen. But first, let's dive into sounds.

Analogue Soul

This is, I'll say up front, a great drum machine in general, but it's the analogue piece that's caused most excitement. The TR-1000 has 16 synth circuits taken from the 808 and 909, skewed towards the 808 as the 909 had some sample-based sounds. Although channels are named on the panel, any of the analogue sounds can be used on any track, but only once per kit of course.

Let's zoom in on the 808 kick as it's so iconic. The TR-1000 808 kick has the same two shaping parameters as the original: Tone, which affects the transient click; and Decay, which adjusts the length of the subby boom that accompanies it. The 1000 version also gets a Tune control.

It doesn't disappoint, delivering that air-shaking thump.

The decay can go way longer than the original. In fact once you get to about 70 percent the circuit oscillates indefinitely, giving you a sub tone that you can shape with the track's envelopes, filter and LFO. This makes sense, as in modern hip-hop production '808 bass' is more associated with sustained bass lines than kick drums. However, as we'll see there's not really a great workflow for melodically sequencing or playing the TR-1000, at least in this first release.

In short, both analogue kicks sound mint, as do all the other analogue circuits. And many have extended capabilities, such as the hand clap with controls for clap size and tail level. Not everything is covered. There's no mid tom, but the other toms can cover its range. Also absent are any of the switched modes on the 808: congas, claves and maracas. For these you'll need to go digital.

A super cool feature of the analogue voices is dynamic responsiveness to velocity, and not just in terms of level. For example, the levels of the Snappy component of the snares and the membrane sound on the toms increase relative to the body tones with harder hits. The TR-1000 has no velocity-sensitive pads, not even the single tap button found on the TR-8S, but it does have the usual Accent system for pushing some steps, and a 'weak note' entry option, giving you three basic levels. It's easy to adjust the velocities of notes in a pattern more finely by holding and dialling in a value. Triggers from MIDI inputs also respect velocity.

Digital Heart

Roland's Analogue Circuit Behaviour DSP was first seen on the TR-8, recreating the 808 and 909 sounds by building the circuits virtually, and staying mostly faithful to the

Roland TR-1000

\$2699

PROS

- It's an analogue 808 and 909...
- ... with many other synth and sample capabilities.
- Much improved file and control workflows.
- Unquantised recording and programming.
- Sample slicing, looping and stem playback.
- Multi-channel analogue and USB audio I/O.

CONS

- No straightforward melodic sequencing.
- Per-step motion has unpredictable results.
- Still no instant takeover mode for patterns.

SUMMARY

A best-of-both-worlds pro drum machine that's hugely fun and inspiring to play.

original controls. ACB has evolved over successive instruments like the Boutique machines, adding sounds from the other TRs and some new and extended control ranges. In fact some sounds like the 6x, 8x and 9x kicks (x being the naming convention for ACB) can be switched between the TR8 and Boutique versions, in either the original or long decay modes that these units offered.

ACB models provide the classic sounds from other great instruments like the 606 and CR-78, as well as 8x and 9x sounds that aren't rebuilt in analogue here. But beyond that, Roland have pimped the ACB patches, unlocking control over many new parameters within the component parts of the algorithms, so they are great to have alongside the analogue versions.

For example, the 808 Snare now has independent control of the levels and decays of the tone (body) part of the sound »



Round the back we have an IEC mains inlet and power switch; USB-C and A ports; MIDI in, out and out 2/thru ports (two of which double up as DIN Sync); a quarter-inch pedal input; 3.5mm sockets for trigger in, filter CV control, trigger out and clock out; two quarter-inch audio inputs; 10 quarter-inch individual audio or trigger outputs; and finally, also on quarter-inch jacks, the main audio outs and dedicated analogue effects outputs. Phew!

» and the 'snappy' noise. The tone circuit is a blend of two basic oscillators, which can now be detuned. The hand clap is even more bendable: you can set the speed and decay of the flammed bursts that approximate a clap, and the tone and decay of the filtered noise that simulates reverb. Perhaps whimsically, the cowbell is completely deconstructed and mod-able. It's fascinating to see the workings of these original synth circuits, and to explore new sounds with them.

As well as ACB, the TR-1000 offers FM and virtual analogue sound engines, PCM sounds and samples. The seven FM models are versatile for both drum and tonal sounds, deserving much more than a single sentence... but there's so much more to get to!

The virtual analogue (VA) models are part of the ACB set, and provide oscillators with all the usual subtractive synth waveforms, with pitch envelopes and PWM as well as the other standard track resources of filters, envelopes and effects. A cool use of a dual track could be to make a two-osc synth voice. PCM sounds are a mainstay on Roland instruments, and the TR-1000 has a ton that you can use without using up any of your project's sample slots.

Difference Engine

If you line up the analogue 808 or 909 kicks against the digitally modelled versions in their default states, they sound different. This is largely because the analogue patches start off pitched way down compared to the ACBs. With pitches matched the 808s are very close although the ACB can't do the infinite release. The 909s on the other hand always sound different, with the ACB version having a kind of subtle grind to its tone (that's nice) compared to the purer starting point of the analogue. The TR-1000's version of the 909 has Tune and Pitch control, slightly confusingly named as Tune seems to affect the pitch envelope while Pitch offsets the overall tuning.

Comparing the analogue and DSP Roland drums is an interesting exercise, but I also spent some time listening to original 808s and 909s and comparing the new analogue versions. I don't have the originals at home to experiment with but there's no shortage of examples to be found on YouTube.

Of course all the 40 year-old instruments are slightly different (often modded) and have a lot of miles on them. And that might



■ The 808 kick generator and track modules revealed in the editor app.

explain the slight difference I hear on the 808 front. Taken as a whole, listening to a complete pattern, the TR-1000's 808 sounds very present and vivid. I don't want to say bright because that's not quite it — it sounds new and fresh. Classic 808s have a subtle dustiness or mid scoop to them to my ears. Both are wonderful. With the 909 I'd say it's too close to call.

Fat Channels

Tracks 1-4 on the TR-1000 combine two channels. This seems inspired by the Analog Rytm, which can layer analogue and sample sources on each track. Layering

“While it feels like a real privilege to have all this drum synthesis power in one place, it's the performance and playability factor that does it for me.”

sounds is obviously useful for thickening sounds, combining a transient with a tone, or just experimenting, but the TR-1000's dual tracks can do more than this.

Using the Layer A and B buttons you can manage and edit the two channels partially independently. You can load two sound sources — a mixture of analogue and digital/sample-based, both analogue or both digital. 808 and 909 combined perhaps? Or maybe an FM sound and a PCM, or whatever. The two sides are independent voices with their own filters, amp envelopes and mods. They share the track's mix settings, which include the compressor, track effect, panning and sends. An excellent feature for layering that I don't think I've seen elsewhere is a polarity flip and timing offset on every source.

The double channels have a Mix knob, generally used for blending the two layers, but which can be reassigned like the other knobs. Placing triggers in a double track's sequence plays both sounds, but you can break this link. Holding either of the layer buttons allows you to place step triggers for either sound independently.

I enjoyed layering a punchy kick with a sub with them triggering together for four-on-the-floor, then interspersing some subs for some groove and rumble underneath. I also tried putting closed and open hats on a dual track and programming their steps separately. Other thoughts were to put the same sound on both layers and use the Mix control to switch between two different patterns, or assign to the Morph fader to juggle between loops. Which leads us to...

Samples

We complete our tour of the sound options with the sample engine. There are 2000 or so samples pre-loaded, with 46GB of space for importing or capturing your own. Sample browsing is integrated with the unified sound generator selection process. I really like how you don't have to think about engines or track types: tracks just become the appropriate type depending on what you load in.

You can sample from the external inputs or resample from the internal mix, or any submix of tracks. You can also capture over USB, although the lack of class compliance limits this to computers, not phones or other USB streaming instruments. By the way, external sources are fed into the mix and can access the effects and side-chain, and external input parameters can be

modulated by the master section LFO. I'd love it if you could assign the external inputs as track sources (as on the MC-707) but that's not possible.

Sampling can be triggered manually, with a threshold, or by pattern start. You can also predefine a length in steps, which is great for resampling to a loop. You can trigger sampling during playback and assign the sample to a track without stopping — great! The missing piece is sync'ed sampling during playback — it's so close to the Holy Grail of real-time loop sampling while jamming. One for the firmware wishlist.

Otherwise, the 1000 does a good job supporting loops and long samples, even stems. There's real-time bpm matching with multiple stretch algorithms. Sample playback stops with the transport rather than continuing indefinitely, and there's a Playthru mode that stops a longer sample from resetting to the start every bar. (Although a slight bug was needing to trim a tiny bit off the end to get it to retrigger every pass). One frustration: you have to figure out the bpm of a sample and dial it in manually. Most samplers will at least have a guess at source bpm based on sample length.

More proof that Roland have thrown everything at this machine: there's full-featured slicing functionality. Matching the Digitakt II's recent update, the TR-1000 even has transient detection and editable slice points. Each slice has independent settings for track features like the filter, envelopes and time-stretch. A dedicated Instant Play sub-mode puts slices on the step keys for real-time playback and record, and you can dial in slices per step in the classic TR mode.

Sequencing

Pattern creation follows the familiar workflow from the modern TR and MC boxes. In TR-Rec mode you enter step triggers in the time-honoured fashion. Instant Play turns the step keys into playable triggers, with the first 10 playing the tracks, then 11 and 12 acting as roll (repeat) modifiers. Inst Rec uses this layout for real-time recording (although no metronome!). A huge departure from previous generations is support for unquantised recording and editable micro-timing. The TR sound might have made its name in house, techno and electro, but is as popular now in genres that thrive on more feel and looseness.

Hold a step and you make a range of quick adjustments such as velocity, timing and probability. Visual guides help with dialling in sub-steps (ratchets), and setting triggers to play only on some cycles. Fine resolution timing adjustment is displayed on a horizontal meter; this, I discovered by accident, can be set with the Morph fader and displayed on the step lights. Nice touch!

At the track level you can push or pull each of your pattern's lanes with micro-timing, set shuffle, and choose different directions and lengths. Track Rotate is interesting. I always appreciate a nudge function on a sequencer to move all steps left or right. Rotate does this non-destructively, offsetting the playhead for the track, which you can see running along the step lights out of phase with the main position.

Roland have kept the TR-8S system of pattern Variations triggered from a cluster of eight buttons. Variations are always a maximum of 16 steps, so creating longer patterns requires >>

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» chaining Variations by holding more than one button. While this is different to how most machines work (but the same as Novation's Circuit), my brain seems to find this easier to keep track of than having one long pattern in a patch.

Patterns also have up to four fills, triggered from the Fill button. Like the TR-8S, you have to choose an active fill in the Edit screen, which means you're probably only ever going to use one. I like the momentary fill mode, which punches in while you hold the button. Which brings me to my number one TR-8S complaint not addressed here. There's no instant Variation or pattern change mode. Changes are always queued, which is a shame, especially if you're working with breaks. There is, however, a Step Loop button, which I'll take over a fill system any day.

Finally, the TR-1000 has a proper song mode, where you can create a playlist of patterns and Variations. It's flexible enough to be useful for arrangement and a longer live performance, or simply as a way to chain Variations out of their natural sequence.

Motion

Motion, or parameter automation, in the TR-1000 is much the same as on the

TR-8s and the MC-707. Motion captures movements or settings of panel knobs: that's the knobs on each drum track and the effects controls along the top. You can capture real-time movements, or hold a sequencer step and dial in values pinned to that step.

While per-step motion capture appears to follow Elektron's 'parameter lock' convention it does not behave in the same way at all. In what's quite a common issue on sequencers, dialling in settings on one step does not affect the step in isolation. Steps with no explicit motion get swept along for the ride. If it's the first step to be written, the values will affect the whole pattern. And even when not, parameters will extend to every subsequent step up until the next automated step. So, if you want to write things specifically to one step you have to write them, then go to the next step and reinstate all the previous values.

This is why trying to capture a melodic sequence is problematic. Although many of the sound sources offer semitone pitch settings, programming a tune feels like hitting a moving target as you end up affecting multiple steps. I hope this is revised at some point. In the meantime, there's one feature that does make this easier...

Snapshots don't do what I expected. I imagined Maschine's Snapshot system, which stores and recalls the states of all parameters in a project. Snapshots on the TR-1000 package the settings of a single track into one of 16 slots, accessed from the step keys in another Instant Play mode. In this mode you can play and record a track with multiple variations. Thus you have a workaround for melodic sequencing: store a set of tuned Snapshots then play them into the sequencer. There isn't a way to drop Snapshots into TR-Rec mode though, or select them after the fact like Slices.

Performance

While detailed programming and arrangement is possible on the TR-1000, it lends itself to getting something basic going then messing with it in an improvisational way. In fact, time spent on prep and programming is better invested setting up effects, knob assignments (which can be multi-parameter macros) and the other new performance features than on making a ton of patterns and variations. And it's always simple to throw a group of faders up or down to change things up.

The new star performer is the Morph fader, which can be used to introduce a whole set of sonic changes in one swipe. You might be thinking Octatrack fader, but it's more like the Quick Performance knob on the Analog Rytm. While the Octatrack fader moves between two scenes of absolute parameter values, Morph applies relative offsets to the current control values on the panel. At the right end of the fader's travel no offsets are applied; modulation is applied gradually as you move left.

Sixteen separate Morph configurations can be stored and recalled from the step keys. A fun way to make use of this — and to emulate a Scene-style workflow — is to park the fader fully left, then tap different keys for immediate jumps between sound states. Like Snapshots, the Morph fader is limited to modulating parameters assigned to panel knobs, which is kind of a shame because you might not want to use up a knob just so the Morph can get to it, and there are parameters it can't reach like the Kit mod.

There are two stages of master effects: digital and analogue. The digital slot is interchangeable with effects that will be familiar from other Roland beat machines. One of these is Scatter, the beat repeater and mangler first found on the TR-8, and which is a great candidate for Morph fader assignment. The analogue stage has



■ All 10 channels are split out, and there's a direct out from the analogue effects for a pure analogue path. The only thing missing is a dedicated aux send.



— The panel is similar to the TR-8S, but the display and encoders offer much easier editing.

a multi-mode filter, which you can use for the usual breakdown tricks of cutting the low or high end; and a lovely saturated Drive. Alongside these are master sections for the shared (and not particularly outstanding) reverb and delay send effects.

Modern Conveniences

As well as bringing the workflow and interface up to modern expectations, there's a solid level of digital connectivity and control. Alongside all the analogue audio I/O there's both USB-C for connecting to a computer, and a USB-A port for external devices. The latter can be used for connecting storage devices for sample import, but will also host a class-compliant MIDI controller.

With the driver installed the TR-1000 shows up on your computer as an audio device, streaming in all tracks individually plus the main stereo mix. It also goes above and beyond on the output side. Multi-channel routing back to the TR-1000 provides a stereo connection to the external sampling input (which runs through the effects) but also direct routing to the mix output and all the individual track outs, which could be super useful in a live scenario.

A surprisingly good desktop app gives you comprehensive real-time control and visual feedback. It would be great to see this as a plug-in at some point for sync'ed recall, and maybe even for automation and audio streaming in your DAW. Sample and slice editing is excellent. The app can also be used to transfer samples to the TR-1000. This is a very basic dropbox kind of workflow. A proper sample manager where you could see and sort what's on the machine would be ideal.

Conclusion

The TR-1000 has been several years in the making. You can imagine that Roland really wanted to do something special if they were going to revisit these heritage analogue sounds. They went back to some of the original artists using the TRs, listened to feedback, tested many original and modded instruments, and rebuilt the circuits with modern components. But they clearly didn't want just to build their own clones. The analogue voices are more versatile and controllable, even while they can serve up the OG goodness, and merging them with their modern platform gives you so much more than just a reissued 808 or 909.

The combination of digital synthesis, PCM and a proper sampler is exciting and compelling. And there are all the split outputs, both analogue and digital, that make this a grown-up pro studio or live instrument. This of course adds up and puts a premium price on the TR-1000. But we routinely review flagship synths and audio interfaces in these pages that cost a lot more.

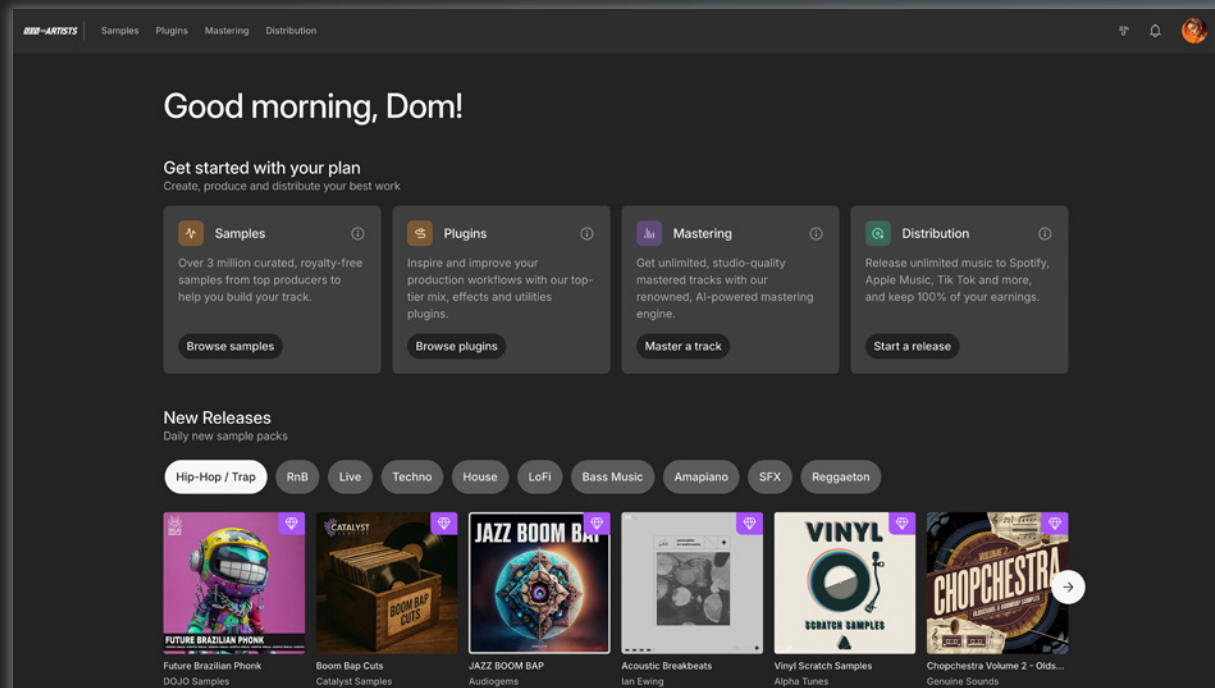
While it feels like a real privilege to have all this drum synthesis power in one place, it's the performance and playability factor that does it for me. The TR-8S has this, but can also be frustrating to interact with when doing anything other than twiddling controls. The TR-1000 has overcome this, and added even more fun with Morphing and Snapshots. All the little ways you can make quick changes add up, and every session with the TR-1000 turns into a jam. It feels like an instrument and in that, as much as in the sound, Roland have recaptured and honoured the spirit of those original genre-defining machines. **///**

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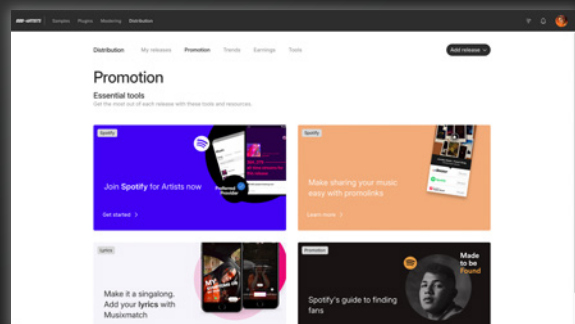
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Zoom's smallest LiveTrak ever is a mobile production powerhouse.

Being only a mosquito's proboscis wider than my Korg Volca Keys synth, the L6 is the most compact LiveTrak I've used, yet into it Zoom have crammed so much, including 10 analogue inputs, two aux outs and MIDI I/O, up to 12-in/4-out USB audio interfacing, and a host of digital facilities. As well as channel EQ, there are also four built-in sampling Sound Pads, a digital effects engine, and a handy 'Scene' facility to store and recall mixer settings. What's more, all this can be powered by four AA batteries or USB, and two USB ports (one dedicated to power, one for interfacing) can deliver power, so there's even power redundancy if needed. While the range of potential applications is huge, you'd expect limitations on any device such as this, and I'll explore those alongside the many plus points. But in case any of it comes across as sounding overly negative, I'll stress up front that overall, I've found the L6 to be hugely impressive, and believe it offers remarkable bang for buck.

Orientation

There's a decent PDF manual, and a software editor makes more possible, but the best way to get your head around this thing is to start by exploring the hardware. You'll find almost everything in terms of I/O and control on the top panel, the exceptions being the battery compartment underneath,

and the USB-C power inlet and microSDXC card slot on the right panel.

The first two input channels are mono, and accept mic/line signals on combi jacks. They share a button for +48V phantom power, and each has a signal-present LED, a mute button and an encoder whose default role is as the channel fader. All onboard recordings are 32-bit float format, but these channels are the only ones to have dual switching A-D converters and, as on similar devices, they have no gain knobs (because they're unnecessary), so are great for beginners or those in a hurry. The flip side is that you have no control over when switching happens, and when it does a short fade is applied to camouflage the change in noise-floor level. This normally works very well, but you may want to double-check the auto-fades on edits made later in your DAW or NLE.

Four more inputs again have a mute button and rotary encoder. These are stereo channels, to give you 10 analogue inputs in total, and accept unbalanced -10dBV line-level sources on quarter-inch TS jacks. They can be used for mono sources too, and the approach is different for each pair of

stereo inputs. Tracks 3 and 4 have a Mono x2 button. When engaged, both inputs appear on the left and right mixer channels, and they share the volume, pan, EQ and send settings — but they're recorded as separate mono files on the SD card. Tracks 5 and 6 instead have a button to select the analogue jacks or a USB return as the input source. The latter are 'hardwired', meaning channel 5 can receive only USB return 1+2, and channel 6 USB return 3+4, so you might find you need to re-patch an analogue source when selecting these. For mono on these tracks, plug into the left jack.

All the input channels feed the master bus automatically (unless you engage their mute). On the right, there are master output jacks at the top, a master output level knob (minus infinity to +20dB) at the bottom, and a stereo LED bar meter between them. There's also a master-bus compressor with automatic make-up gain. This offers no controls besides an on/bypass button, but can work well for raising levels of dialogue or taming filter sweeps during performance/streaming, without sully any multitrack recordings. Top right is the quarter-inch TRS headphone output, also with its own

Zoom LiveTrak L6

\$330

PROS

- Sounds decent.
- Switching 32-bit conversion for the mic preamp channels.
- Incredibly portable, with multiple power options.
- Super-easy to use.
- Great bang for buck.

CONS

- Some routing limitations.
- Software could use more features.

SUMMARY

The L6 packs a frankly insanely good feature set into an immensely portable package! There may be limits to what it can do, but there's much to love.

monitor level knob. This always delivers the master mix to your headphones: there's no separate solo facility, and, short of unplugging them, no way means of muting your speakers if recording vocals while monitoring on headphones. Above the headphone monitor level pot are four buttons: Record and Play for the multitrack recorder, and the three Scene buttons.

Immediately to the right of the main input channels, a profusion of buttons set the function of the channel encoders, and the selected function applies to all channel encoders simultaneously. In addition to the default level function are pan, the effects send levels, aux 1 and 2 sends, and four buttons for the channel EQ (comprising two shelves and a sweepable mid). Hit a button and the LED rings around the encoders will indicate that parameter's current settings. There's more you can do with the software, but there's a decent amount of physical control for such a small device. My only gripe is that the LED rings can sometimes be obscured by the encoder knobs, and the light from the LEDs can, from some angles, appear blurry, making it hard to be certain of the setting.

Sends & Scenes

To the right of these buttons are more I/O and a couple of internal channels. At the top are MIDI in and out on TRS mini-jacks, and the audio interfacing USB port. Immediately to their right are two aux sends on quarter-inch jacks. Note there are no dedicated aux return channels (so external effects must occupy input channels), and that the aux

sends are mono even from stereo channels, which won't suit everyone (eg. if trying to incorporate a stereo sampler into a live rig).

The built-in digital effects may be rudimentary but sound decent enough. A Sel button toggles between three reverbs (Hall, Room and Spring) and two delays (Delay and Echo). When controlled from the hardware, these are pretty much on-or-off affairs, but there's a tap-tempo button for the delays (tempo will also slave to a MIDI Clock signal), and a pot to set the overall effects level. The final channel, to the left of EFX, is for the Sound Pads, of which more below.

To save a Scene, you just hold and press one of the three buttons, and press again to recall. Most parameters are covered, including some that can only be set using the software editor. For example, it will store the channel levels and mute states, and the EQ settings. So you could prepare Scenes in advance and recall them during a show, for instance one Scene to solo a podcast host by muting only the guest mics, and another to reverse that. Something it doesn't do is recall the encoder function that was 'in view' when saving; if that's important you'll have to select it manually.

Multitrack Recording & Sound Pads

To make a multitrack recording, pop your microSD card in the slot (extracting it again proved unusually fiddly!), set the desired channel levels and press Record. When you're done, hit Play/Stop. That's it! The multitrack comprises mono files for channels 1+2, mono or stereo files for channels 3-6 (depending on how you configured them), and the stereo mix, complete with the effects and Sound Pads. All files are

USB Audio Interfacing

The L6 worked as a USB audio interface straight off the bat with my Mac and my Samsung Android phone, but Windows users need to install a driver. The 48kHz sample rate is fixed, but you can choose a 32-bit float or 24-bit fixed-point word length. As well as recording to and playing back from your DAW, you can easily use a phone to play a backing track into the mixer during a live performance. When I received the L6, it had v1.0 firmware installed, and updating it to v1.2 introduced new features. Notably, button combinations during power-on now configure how the L6 presents itself to attached devices. By default, the host determines how many I/O are used, but you can choose a fixed 12-in/4-out format, or a simpler 2x2 stereo configuration. The latter is particularly welcome: it means it's now possible to hook the L6 up directly over USB to my DJI Osmo Pocket 3 video camera! So I can record a mix in sync with the video, and (albeit without any timecode support) pretty easily line up any SD multitracks in post.

48kHz/32-bit WAVs. Playback is equally simple: hit Play, and the last recording plays back.

For a budget device, it sounds beautifully clean and detailed, but I would ideally have liked more features. For instance, using the hardware you can only play back the last-captured recording; there's no way to access previous takes. Also there are no transport, marker or other file-navigation facilities, and there's seemingly no way to rebalance the multitracks on playback. Finally, during playback you cannot monitor the input channels, and you can't overdub. These are the sort of limitations I had in »

■ The LiveTrak L6 has a dedicated USB power inlet on the side, but it can also be powered by AA batteries or via the other USB port on the top — so you have plenty of options for portability and redundancy.

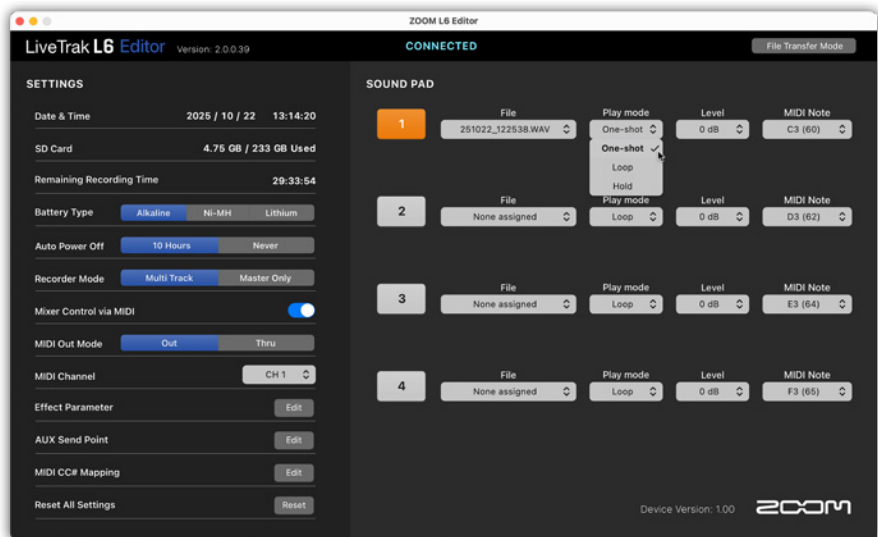


» mind at the top of this review — while they will, I'm sure, put off some potential users, most probably won't see any problem here.

Recording to the Sound Pads is almost as easy: hold down the main Record button, hit the target Sound Pad, and it will almost instantly start sampling the stereo mix. Hit the pad again to stop. Thereafter, hitting the pad triggers the captured sound. You do need to get your timings spot-on when recording to avoid unwanted sound/silence at the start of the sample, since there's no built-in sample editing. But you soon get the hang, and can always record a new sample to the pad (the previous recording remains on the SD card, but can't be accessed without a computer). Note that you can't send the Sound Pads to the effects or auxes, there's no access to EQ or pan, and when using the hardware alone you can't adjust the individual pads' playback levels, or delete a recorded file...

Software & MIDI

Which brings us neatly to the LiveTrak L6 Editor software! Installation was quick and easy, despite my macOS v12.4 not officially being supported. When connected over USB, the software accesses various functions unavailable on the hardware. It can set the L6 to file transfer mode too, making it a card reader: you can use your OS file browser to move/copy files, or third-party software to edit Sound Pad files. You can also load new Sound Pad samples onto the SD card (pre-recorded segments to trigger during a podcast show, for instance, or a backing rhythm track for busking). The file names for the onboard



■ The LiveTrak 6 Editor software offers a range of useful functions, from managing the Sound Pads to changing the aux send point, tweaking the effects parameters and assigning MIDI control.

recordings have a date/time format, and until I'd connected the L6 to the software that date/time wasn't, erm... up to date! So it's worth hooking it up once before you do any serious recording.

When not in file transfer mode, the software lets you tweak or assign MIDI control for various functions, including more than you can access on the hardware. For example, you can set up direct MIDI control of the individual EQ bands and aux sends, and can assign different MIDI controls to those functions for different channels. You can also do things like set the L6 to record the master mix only (saving card space/increasing the available recording time), change the aux send points (pre- or post-fader on a per-channel

basis), and access additional basic effects parameters. Usefully, while that last one can't be done using the hardware controls, the settings can be saved/recalled using the Scene buttons.

A window for the Sound Pad files accesses more useful functionality. You can select which of a pad's stored files is loaded, set each pad's playback behaviour (one-shot plays through the full sample, loop does the same repeatedly, while hold gives the pad a momentary action, the sample restarting each time it's pressed) and, importantly, adjust the playback level of individual pads, using a drop-down menu (-10 to +15dB in 1dB steps). You can even map MIDI notes to trigger the pads' samples, which could be handy. Yet again, what's here is decent, but I feel more could be possible. Basic sample-editing tools (trim, fade and gain), for instance, would be really convenient, as would a MIDI Learn feature for attached devices.

The Bottom Line

As I said above, though, the L6 can't be all things to all people, and I'm sure we could all write very different feature wish lists! But I can think of nothing else that offers so much in such a compact form, for so keen a price. It's hugely impressive, fun and really immediate to use. For some, the L6 could be the only preamp, recorder or interface they need. For others it might prove an attractive backup tool for the kit bag. ■■■



■ There are two aux sends, which can be set to operate pre- or post-fader.

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Audio Interface

Grace Design m701

Grace Design have brought their considerable expertise to the audio interface arena.

HUGH ROBJOHNS

Grace Design are well known as purveyors of some of the very best microphone preamps and monitor controllers available, but there are more strings to the company's bow, with highly regarded instrument preamplifiers, active DI boxes and microphone mounting systems in the portfolio — all brilliantly conceived and beautifully engineered, too. However, the latest addition to the family is a first for Grace Design: a sophisticated audio interface and A-D/D-A converter!

This new product, the m701, features a fully modular construction allowing the user to specify their own specific analogue and digital I/O requirements, with up to 64 channels of analogue conversion (in or out) and 212 digital channels in a wide variety of formats. The way that is arranged is through eight slots housing a mix of eight-channel balanced analogue input and output cards (or four-channel mic preamp cards), plus two slots for 32x32 digital cards (in AoIP, HDX or USB flavours), and a further option module currently used for 24 extra channels of AES3 — all this

in addition to S/PDIF, ADAT and eight channels of AES3 I/O fitted in the unit as the base standard.

Although Grace Design have never offered a fully-fledged interface before, digital conversion is certainly something of which the company have plenty of experience. After all, they have been incorporating high-performance converters into their mic preamps and monitor controllers for 25 years or more, allowing Grace's engineers to build up all the knowledge and skills needed to extract the very best technical performance from high-end converters — expertise in analogue stages, power supplies, clock generation, grounding arrangements and much more, where small design tweaks have a big impact on the technical specifications.

Overview

The m701 is a 2U rackmounting unit with a typically understated grey chassis and front panel, dominated by a large (WVGA) colour display screen, a rotary data encoder, six menu navigation buttons, and an illuminated standby/on button. It is a simple, elegant user interface which belies the massive connectivity and

signal routing potential hidden within. The standby button illuminates yellow in standby mode, green while booting, and white when operational.

Not surprisingly, the rear panel is a lot busier, being completely awash with connectors, many on interchangeable cards or modules to facilitate individual configuration requirements. Starting at the left-hand side, a universal IEC mains inlet accepts 100-240 V AC and has an integral mains on/off switch and fuse holder. Other connectivity fitted to the chassis as standard includes a single Ethernet RJ45 port (for network control via GraceNet — a browser-based remote-control app), and a USB-A host socket for firmware updates or a Wi-Fi access point connection. Additionally, standard digital audio I/O is provided on a pair of Toslink optical sockets, an AES59 (25-way D-Sub) socket for eight channels of AES3, a pair of RCA phono sockets for coaxial S/PDIF, and a couple of 75Ω BNCs for word clock (with switchable termination on the input). The optical ports can be configured individually either for dual-channel S/PDIF (operating at base and double sample rates) or ADAT, the latter carrying eight channels at base

sample rates or four channels (using S/MUX) at double sample rates.

All other connectivity is configurable, starting over on the right-hand side with eight vertical module slots, which can be fitted with any desired combination of eight-channel balanced analogue line-level inputs, eight-channel balanced analogue line-level outputs, or four-channel mic/line modules, as mentioned earlier. All connections to the outside world are via AES59 (25-way D-Sub) sockets mounted on the individual modules.

The four-channel mic cards use the same preamp design as found in the company's superb m108 and m801 MkII preamps, and if two mic cards are installed alongside each other a ribbon cable links them so that all eight inputs are conveniently accessed through the first card's AES59 connector. In situations requiring additional (or external) mic preamps, up to 12 external m108 microphone preamplifiers can be connected to the m701 over Dante, and fully controlled through the GraceNet user interface, with all settings stored within the m701's memories.

A further two horizontal 'Digital Interface' slots on the rear panel each provide up to 32 more channels in and out. Compatible interface cards are currently available for Dante, Ravenna (and AES67) and Pro Tools DigiLink, and a USB 2.0 interface card should also be available by the time you read this review.

Any pair of interface cards can be installed, although the Ravenna card (which uses the Merging Audio ZMAN networking hardware) can only be used in slot 2. The Dante card employs the Brooklyn 3 networking hardware, and although it supports 32 channels in both directions at base and double sample rates, this reduces to 16 channels at quad sample rates. The Dante and Ravenna cards are both equipped with dual RJ45 sockets for primary and secondary network connections. For Pro Tools HDX/HD and HD-Native users, the DigiLink interface effectively emulates two Avid HD I/O interfaces. Again the card is physically equipped with two sockets (Primary and Expansion), but only the Primary socket is currently enabled.

A third, much larger module slot accepts an optional panel carrying three more D-Sub sockets for an additional 24 channels of AES3 in and out. As with the standard built-in AES3 port, each of

these eight-channel interfaces can be configured individually for professional or consumer status data formats.

The m701 operates with a single clock domain, which means that all digital inputs must have the same sample rate and be synchronous with each other — ie. all clocked from the same master clock source. Anything that is not at the selected sample rate or correctly synchronised with it is automatically muted. There are no sample-rate conversion facilities in the unit with the current firmware or, indeed, any kind of signal processing capability such as level adjustment, EQ, Mid-Sides encode/decode — although the built-in monitor mixer does have channel faders, so there is some DSP capability available within the FPGA. Perhaps Grace Design plan to expand the feature set in the future, but they have announced no plans so far.

Assuming the synchronous clocking requirements are met, any physical input can be routed to any number of physical outputs, configured using a simple cross-point matrix graphical interface, with input sources arrayed on the left-hand edge of the grid, and output destinations

Grace Design m701

From \$2995

PROS

- Comprehensive range of digital and analogue I/O options.
- Massively configurable.
- Typically elegant and practical user interface.
- Impressive versatility and technical performance.

CONS

- Mic input cards only accommodate four channels per slot.

SUMMARY

The Grace Design m701 is a cleverly conceived high-end interface/converter unit with a wide range of interfacing formats and a high channel-count capability that makes it ideally suited to larger studio and mastering installations. Nicely engineered, and with excellent technical specifications, the m701 is an impressive product.

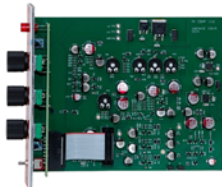
across the top. Channels can be routed individually by clicking on the appropriate cross points, or (by double-clicking the Edit knob) as a complete block, with all source inputs within the selected interface module being mapped directly to their

»

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-Jason Kingsland
(Belle and Sebastian, Band of Horses, Deerhunter)



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» corresponding destination module outputs — a major time-saver. Similarly, existing routings can be erased individually or across a complete block.

I mentioned the inclusion of a monitoring mixer just now, and there is a 32x8-channel low-latency mixer built in specifically to allow independent monitoring of any input or output. The control screen looks much like any standard DAW mixer, with familiar functions like mute and solo buttons, faders, pan knobs and input and output routing boxes (with eight groups). The eight monitor mix output busses can be used as individual mono sends or combined in odd-even pairs to create stereo sends, and routed to any physical output. A really neat feature is that the m701 can generate a QR code on screen for each monitor mixer output, allowing mobile devices to connect over the network to the relevant channel very quickly and easily!

Naturally, each of the installed interface cards can be configured as required through dedicated menu pages. For example, on the analogue input and mic input cards, the converter's anti-alias filter characteristics can be selected from four different options: fast roll-off linear or minimum phase, or slow roll-off linear or minimum phase. Minimum-phase selections have the least latency, of course (both are five samples), while the linear options are longer at seven samples for the slow roll-off and 19 samples for the fast roll-off.

The input sensitivity of the analogue line-level input cards can also be switched for 0dBFS to equate with +18dBu or +24dBu (the EBU and SMPTE standard alignments, respectively). But there is also a ± 20 dB digital trim control (with 0.1dB increments) for meeting pretty much any system's calibration requirements.

Technical Performance

Entirely as expected, the technical specifications for the m701 are completely beyond reproach. For example, the mic preamp cards have an EIN figure of -127dB (150 Ω source and 60dB gain) with a maximum input level of +30dBu (at the -6dB gain setting). Matching between channels is within 0.05dB and crosstalk is -122dB (at 10kHz). I also note that the A-D input cards have a fixed high-pass filter at 15Hz, whereas the D-A output cards are DC-coupled. And when it comes to saving the planet, the unit consumes a maximum of just 55 Watts, which is impressive considering what's going on inside. It is fan-cooled (at the right-hand side), but it ran extremely quietly in the whole time I used it.

When it comes to the analogue I/O cards, at a 48kHz sample rate the overall latency from an analogue input to any digital output is just 0.2ms (with a minimum-phase filter), rising to 0.5ms with a linear-phase filter option. For any digital input to an analogue output the latency is the same with a minimum-phase filter (0.2ms) but slightly longer with the linear-phase options (0.8ms). Obviously, these times all halve with each doubling of sample rate.

I ran my usual AES17 dynamic range tests, achieving figures of 121.1dB (A-weighted) for the A-D card and 126.7dB (A-weighted) for the D-A card, which are superb results, both improving on the published specs, too (120dB and 125dB, both A-weighted, respectively). These are exceptional results, ranking the m701 second in my table of measured D-A converters and seventh in my table of measured A-D converters. And at the top of these tables fractions of a decibel can dramatically change the rankings!

More impressively, the Grace Design m701 outperformed Prism Sound's Dream ADA-128, by over 2dB in both measurements, and sits comfortably between Merging Audio's Hapi AoIP interface and the table-topping Lynx Hilo 2. Confirming this sublime level of technical performance, the THD+N figures were equally impressive at 0.0002% and 0.00055% for the A-D and D-A cards, respectively.

When mic modules are installed in the m701 additional configuration features become available, including phantom power on/off, polarity inversion, adjustable gain (+2 to +69 dB in 1dB steps, plus a -6dB setting), and a ribbon mode. This last option not only disables phantom power, but also increases the input impedance from 8k Ω to 20k Ω to improve the low-frequency and transient response of ribbon mics. The -6dB sensitivity option allows the mic input to accommodate up to +30dBu, and the 20k Ω input impedance is sufficiently high for most line level devices, too.

The analogue line-level output cards have the same reference alignment and digital trim options, but six different reconstruction filter modes. In addition to the linear- and minimum-phase filters, and fast and slow roll-off options, there are hybrid and apodizing filter modes, too. The minimum-phase slow-roll-off

filter has the lowest latency at just 3.5 samples, while the linear-phase fast-roll-off and apodizing filters have the highest at 35 samples.

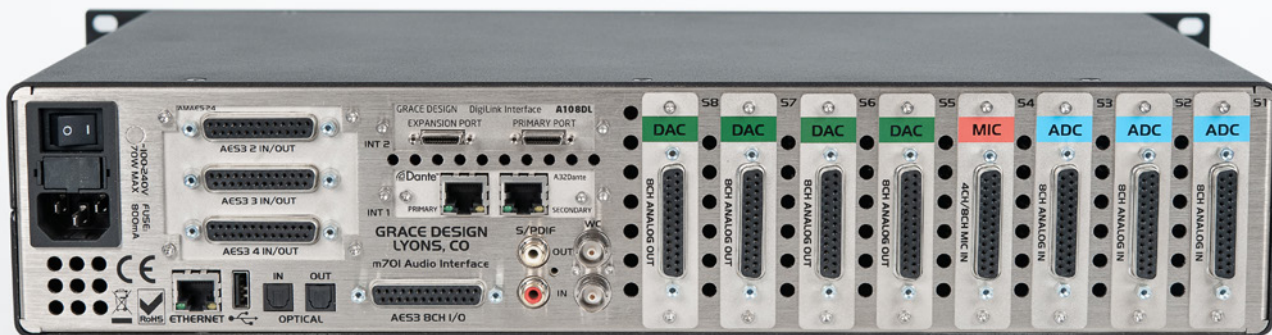
Apodizing filters are very clever because they remove the filter-ringing artefacts inherited from any and all upstream conversion stages, while the hybrid filter is a halfway house between the linear- and minimum-phase fast-roll-off filters: almost as steep as the linear-phase filter, but with mostly post-ringing and less latency.

Operations

Setting up and controlling the m701 is entirely logical and straightforward, even without consulting the well-written user manual. I had no surprises when incorporating the review unit into my own studio.

Everything can be controlled directly from the front panel or, as the unit

»



■ In contrast to the calm of the front panel, there's a lot going on at the rear of the m701. The eight vertical slots [S1-S8] can take any combination of eight-channel line input [A-D], eight-channel line output [D-A], or four-channel mic input cards, to suit the user's requirements. The two horizontal slots to the left of these take the high channel-count digital I/O cards — any two from Dante, Ravenna, DigiLink or USB. Currently, the leftmost slot can be filled with 24 channels of AES3 (as shown) or left blank.

D3V



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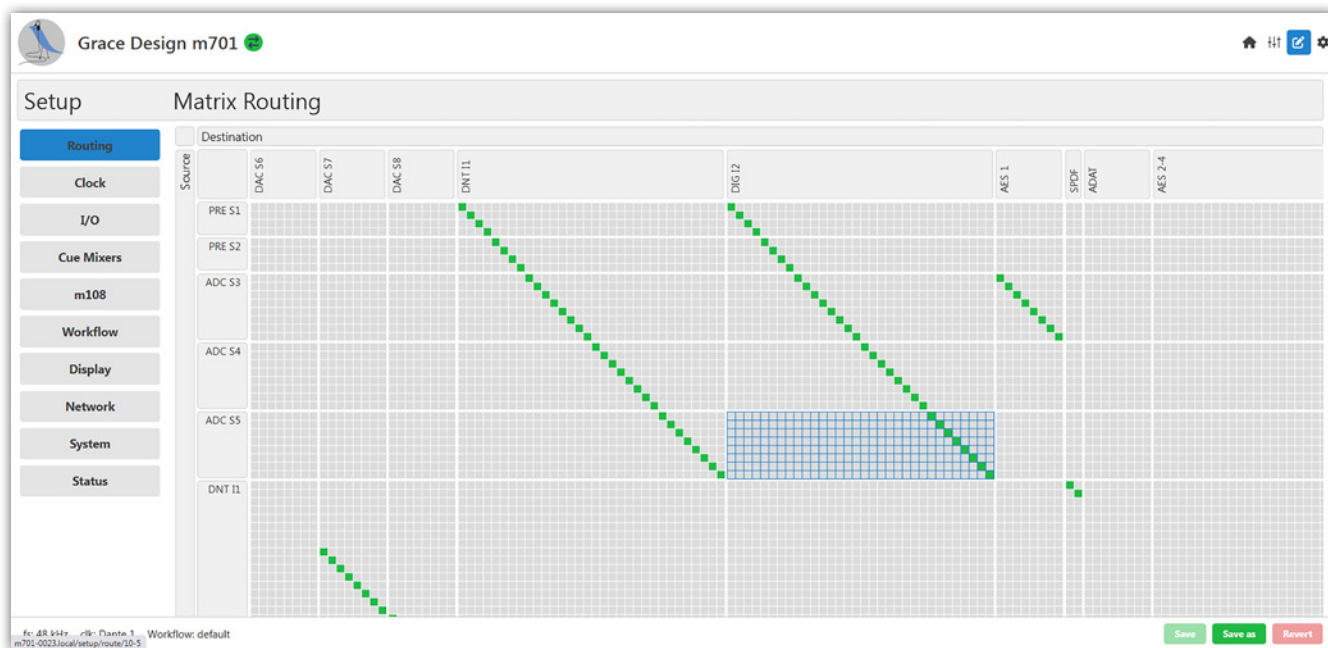
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The m701 can be controlled from its own screen, or remotely from a web browser (as shown here), with identical options and screen layouts. This Routing page shows the cross-point matrix enabling any source (left edge) to be routed to any number of destinations (top). The green squares show confirmed routing cross points.

» incorporates its own web server, via the GraceNet interface, which is accessible from any web browser. When connected to a local area network with DHCP, the web-access address is shown in the lower right-hand corner of the display in the form <http://m701-xxxx.local>, where the 'xxxx' is part of the unit's serial number. Entering this address into a web browser instantly accesses the unit remotely. (A static IP address can also be allocated, if preferred.)

The unit's menu system defaults to a home screen showing signal levels on vertical bar graphs (analogue I/O above digital I/O), with function tabs across the top and status data (sample rate, clock source and IP address) along the bottom. The navigation buttons or encoder knob allow different menu tabs to be highlighted — Inputs, Outputs, or (monitor) Mixer — and pushing the encoder then opens the selected tab. If viewed via GraceNet all inputs and outputs can be seen on the same screen, which is an advantage on a fully loaded system.

Configuring the unit is done via the Setup menu, accessed by pressing a navigation button to the right of the screen, marked with a cog symbol. This opens a dedicated menu with pages for Routing, Clock, I/O, Workflow,

Display, Network, System, Status and Factory. A Back button exits the selected menu, and if any settings have been changed a dialogue appears to confirm or cancel the changes.

The functions of most of these menu pages is obvious: the Routing page displays the routing matrix as described earlier, and the Clock page selects the desired internal or external digital clock source. Options here include the two interface modules, AES3, optical port, S/PDIF, word clock or the three extra AES inputs (if that module is installed). There are also options for terminating the word

clock input, and for selecting what the word clock output socket sends (either a loop-through of whatever is connected to the word clock input, or the internal system's clock). All standard sample rates from 44.1 to 192 kHz are supported.

On the I/O menu, specific configuration options are presented for the installed A-D, D-A or mic preamp cards, as well as the digital interface modules and the standard-fit digital connections. There are settings for the converter filters, phantom power, gain, trim and reference levels, and so forth, all as mentioned earlier.

The Workflow page may sound unfamiliar, but it is essentially where all

user settings and system configurations are loaded, stored, copied, named and imported or exported. Again, all very straightforward and logical. The Display page handles functions like the meter peak-hold duration, the front-panel normal and dim brightness levels, and the dim timeout duration — the display automatically dims a set time after the front panel controls were last used (between 10 minutes and an hour, or no dimming).

“It has surpassed even my expectations in terms of configurability, ease of use and technical performance.”

Pricing

Modular Components

- m701 Base Unit \$3150.
- m701 8-channel A-D and D-A cards \$895.
- m701 4-channel mic preamp card \$895.
- m701 Dante or DigiLink card \$895.
- m701 24-channel AES I/O card \$750.
- m701 USB 2.0 card \$695.

Factory Configurations

- m701 8x8 (DigiLink or Dante): Base Unit with eight channels A-D, eight channels D-A and DigiLink or Dante card, \$5543.
- m701 16x16 (DigiLink or Dante): Base Unit with 16 channels A-D, 16 channels D-A and DigiLink or Dante card, \$7244.
- m701 32x32 (DigiLink or Dante): Base Unit with 32 channels A-D, 32 channels D-A and DigiLink or Dante card, \$10,645.
- m701 Production (DigiLink or Dante): Base Unit with 16 channels A-D, 16 channels D-A, eight-channel mic pre and DigiLink or Dante card, \$9134.

A Network page does exactly what you imagine it would, with options for setting DHCP or a static IP, gateway address, host name, and so on, while the System page accesses information including serial number, firmware revision, GraceNet version, system update options, debug logs, and digital interface card setup functions (for Dante and Ravenna network options etc). An internal SD memory card is used for storing logs and other user data, and there's even a menu function to test the SD card's health.

The Status menu provides information on the internal power rails, temperature of the main circuit board, and cooling fan speed, while the Factory menu — which requires a passcode to enter — is where the factory programs the unit's serial number and FPGA, and it is not normally accessed by the user. The unit performs a self-test on boot up and if it detects any changes to the hardware (eg. modules added or removed) it warns the user and automatically modifies the last used workflow to take into account any missing or new routing options and configurations.

Setting the unit up via the front panel display and controls is remarkably easy, but the size of the screen inevitably means there's a fair bit of scrolling around — especially for routing. Using the GraceNet web interface generally allows all options to be seen at once (depending on the size of the computer screen), and I found I tended to use that more often than the front panel — although the meter screen was very useful for checking signal routings at a glance. Either way, the overall look and feel is very consistent between the two user interfaces and I had no trouble working with either.

Impressions

Grace Design's new m701 feels much like their other products in the ways its menus are structured and settings configured. The range of I/O modules is impressive and, despite the huge I/O count, establishing connections between inputs and outputs is very straightforward and logical, as is saving and recalling different setups.

Having reviewed the Prism Sound Dream ADA-128 in December 2023 I was naturally drawn to comparing the two units, which share broadly similar design goals, applications and I/O options. It's fair to say that, in terms of quality and capability, the ADA-128 has a substantially larger routing capacity, and its four independent clock domains make it far more versatile in complex multi-room mastering situations — although that would be a very specialised requirement that most users of this kind of interface probably won't require. Technically, the m701 slightly outperformed the Dream ADA-128, which is very impressive indeed (see box).

When I heard the announcement of the m701 I thought it was a logical addition to the Grace Design portfolio, and it was bound to be an impressive performer. However, it has surpassed even my expectations in terms of configurability, ease of use and technical performance. This is one heck of a big-boy's interface! **///**

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PAUL WHITE

Like its predecessor, the BeatBuddy 2 is essentially a drum machine in a pedal, and it's designed to support your live performance — but it also makes a far more friendly alternative to a metronome for practise or recording purposes. It's available in two versions: the full-fat BeatBuddy 2 (which I'll abbreviate to BB2) that's reviewed here, and the more keenly priced BeatBuddy Mini 2 (BBM2). The latter, of course, forgoes some of the full version's features, and I've written more about that model in the separate box.

Among the headline new features in the BB2 are 'HD' drum kits — a definite step up in quality from the original. Singular Sound have also created a 'song matching' tool, that the user can use to search for famous songs and discover what beat and tempo should work. That can be found at <https://songmatcher.singularsound.com>, and note that songs that are listed as compatible with the BB1 will also work for the BB2. The BB2's output is stereo, and for best sound quality the pedal should be connected to a full-range, flat-response system such as a PA system or keyboard amp. It will still

Singular Sound BeatBuddy 2

Rhythm Generator Pedal

Singular Sound's revamped drum generator offers top-quality sound and flexible playback options.

work in mono if plugged into a guitar amp, but of course the sound quality will be affected by the amp's own character.

Overview

Physically, the BB2 looks rather imposing: while it comes in a compact pedal format, it's solid and chunky enough to remain stable on the floor during use. It's operated by two domed footswitches, three turn/push encoders (these select the drum set, tempo and volume), and another multifunction encoder comprising a Tap button surrounded by a left/right, up/down ring. Pressing Tap brings up Tap Tempo

mode, but it's also used as an Enter button when using the directional pads to select functions, including the genre, tempo, content menu, and song folder.

A large colour LCD displays the selected kit, tempo, time signature and other useful information. The status for Intro/Outro, Drum Fill and Transition is also displayed clearly, with a moving block display that indicates the progress of the beat, and good use is made of colour to show intros, outros, main beat, transitions and so on. The switches recognise press, multiple press and hold gestures, which means you

Singular Sound BeatBuddy 2

\$499

PROS

- Robust and stable pedal format.
- Choice of well-sampled drum kits.
- Comprehensive controllability.
- Manager editor allows content creation on a computer.
- Clear, informative display

CONS

- No USB means the SD card must be removed to edit content.

SUMMARY

A substantial improvement over the original BeatBuddy, this pedal makes it easy to tailor your drum backing to your live playing or practice needs.

can do things like call up an outro part on the fly, allowing for the kind of leeway in live performance that a pre-recorded backing track can't give you.

As the footswitches can handle multiple functions, the display also tells you what will happen if either footswitch is pressed. For example, if you hold down the left footswitch to bring in a transition fill, the word Transition appears in the display. Likewise, if you do a double-press, the display changes to Outro and the background colour changes to red leaving you in no doubt as to what is happening.

The cast metal case has various jack connections on either side. On the right are two Multiswitch footswitch/expression pedal jacks, plus left (mono) and right inputs that allow a mono or stereo signal to pass through the pedal unaffected by its volume control (which governs only the level of the generated drum parts). This is useful if you don't have a separate mixer. Note that the user must set the type of accessory being used with the Multiswitch jacks in the Multiswitch settings menu, which is accessed by pressing the drum

set and tempo knobs down at the same time. On the left side, you'll find left (mono) and right output jacks, along with a quarter-inch TRS headphone output jack.

On the rear, there's an input for a 9V, centre-negative PSU (a 500mA power supply is included). There are also five-pin MIDI in and out sockets here, as well as an SD card slot. A 4GB SD card is included and all the drum kits, beats and settings are stored on this, so it must be inserted before the pedal will work. If the SD card is lost or damaged, or you want to keep a backup, the content can be downloaded to a fresh card from the support section of the Singular Sound website.

The way that the BB2 works is that the SD card stores two categories of content: the sampled drum kits themselves, and the Song files. Songs are made up from MIDI files that are organised into separate song sections. Drum sets use a WAV file format, and these are triggered by the Song MIDI files. Using the web-based BeatBuddy Manager software, users can create their own songs, design new beats and import their own drum sounds. BeatBuddy 2 can also be used as a drum sound module using MIDI from a drum pad controller or similar, and MIDI files of beats created using DAW software can also be imported. The majority of BeatBuddy 2's drum sets comply with General MIDI mapping.

It should be obvious that this approach allows the BB2 to play beats at any tempo without compromising the audio quality, but it's also worth pointing out that it makes it possible to play back the same sounds in any time signature — something you can't really do with audio loops. On board are 15 drum kits spanning 24 genres, each offering numerous variations along with intro, outro and fills. Apparently, the BB2 can read the SD card from the BB1, which could be handy if you're

upgrading, but it won't work the other way around. While the drum rhythms may not rival high-end sample libraries for nuance, I have to say that the sound quality is excellent, and everything sits well in a live performance context. If a PA feed is not available, it still sounds surprisingly solid when played through a guitar amplifier, though it's definitely worth playing it through a decent PA/keyboard speaker system if at all possible.

The Buddy System

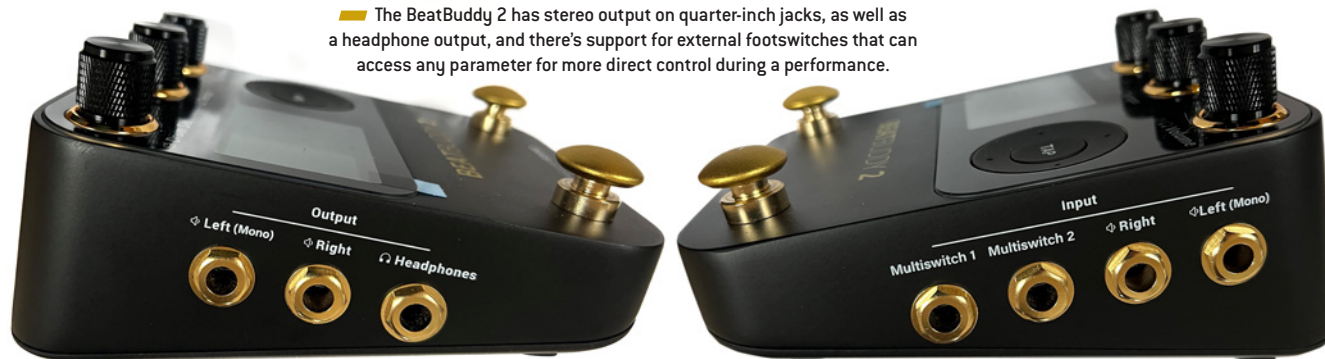
A press/click on the Volume encoder toggles its turn action between controlling the main output volume and the headphone volume. And once you can hear what you're doing, it only takes a little practice to get used to triggering fills and (with a double-press), outros, or switching song sections. Indeed, because whatever you trigger happens at the start of the next bar or phrase, I find it rather easier than using many loopers (where accurate timing is essential).

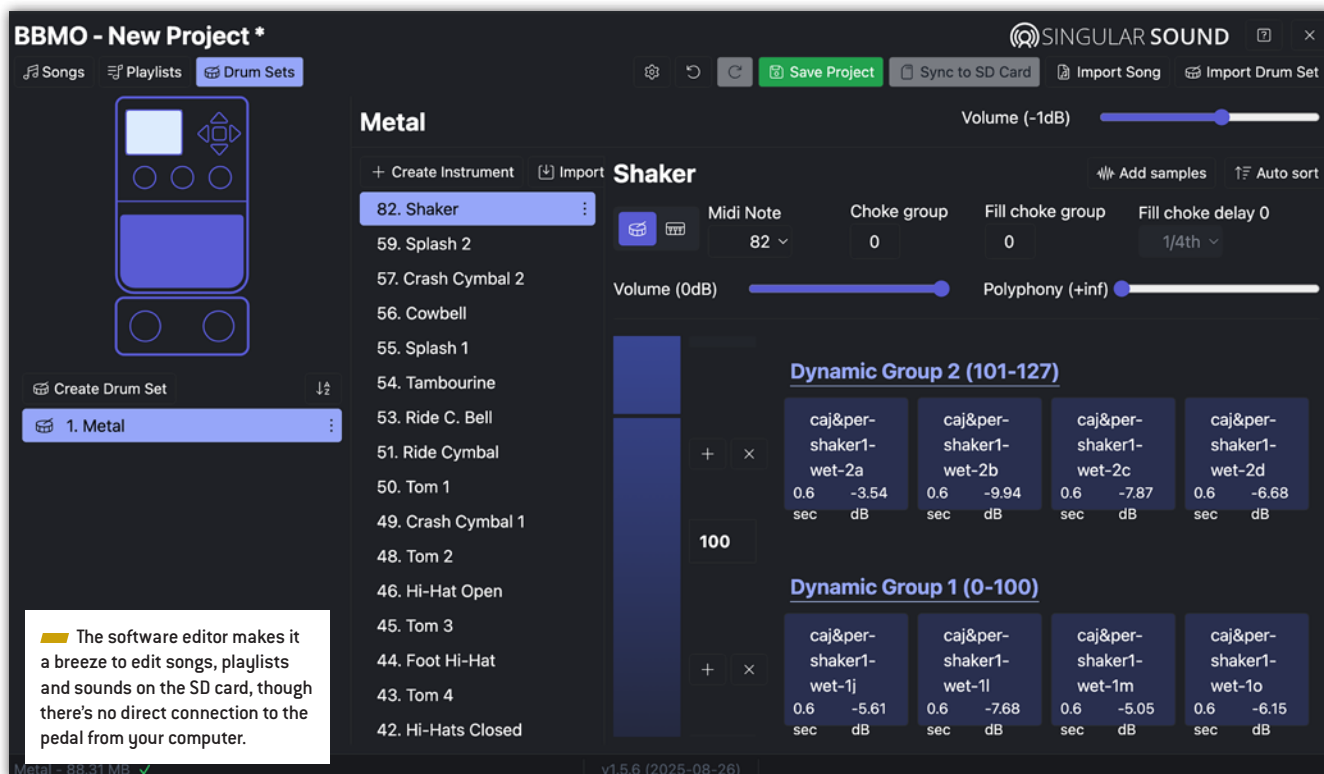
That will be all many users require — and there's an argument for keeping it simple — but to cater for those who want more control there's an optional Superswitch floor unit, whose six domed footswitches can give you direct access to all the key functions. This allows for things like hands-free triggering of accents, moving to the next or previous part, doubling or halving the tempo, and navigating the song folder. And if six extra switches seems like overkill, there's also a two-footswitch option.

Choosing a drum kit is straightforward: you just use the Drum Set knob to scroll through the options, then press the encoder to load it, which takes just a few seconds. Pressing and holding will save that kit as the default for the current song. The tempo can be set using the Tempo knob or by using Tap Tempo mode, and

»

■ The BeatBuddy 2 has stereo output on quarter-inch jacks, as well as a headphone output, and there's support for external footswitches that can access any parameter for more direct control during a performance.





» the BB2 can also sync to MIDI Clock. As the controls are moved the display automatically switches to the appropriate readout until the adjustment stops. And since I mentioned tempo, a feature unique to the BB2 is the ability to connect an expression pedal for control of volume, tempo and playing dynamics, for a more convincing 'performance'.

Once a suitable drum kit, beat and tempo have been selected, the display shows large blocks tracking across the screen to match the set tempo, providing a good visual cue to get you off to a tight start. If you stay with the default settings, pushing the left footswitch plays an optional intro, followed by the Part 1 beat. Tap again to get a fill, which will vary each time. Holding down the switch initiates

a transition beat, and when released the song moves on to its next part. A double-tap ends the song, which may have an optional outro fill just to round things off. Pushing the right footswitch plays Part 1 without an intro, and pressing it again puts the machine into pause mode. Press again and you unpause. Using external switches does make it easier to navigate the song folder and play accented hits and so on, but by no means are they essential; I could do everything I needed with the BB2 alone.

One big advantage of the full-fat model over the Mini is that with the former, the BeatBuddy Manager web app facilitates the organisation and editing of drum parts and also allows the use of content purchased from the Beat Boutique library

(www.singularsound.com/pages/beat-boutique) or downloaded from the user forum. You can also create your own rhythms and drum sets from there. In case your current browser doesn't play nicely with BeatBuddy Manager (Apple's Safari is not officially supported), I can confirm that Google Chrome worked on my Mac, and other listed supported

BeatBuddy Mini 2

The BeatBuddy Mini 2 omits the standard BeatBuddy 2's MIDI functions, as well as its card slot, HD drum sounds, Autopilot feature and expression pedal-control of volume, tempo and playing dynamics. In terms of connectivity, there's just the power inlet, the main I/O jacks and a footswitch jack. Also worth noting is that the BeatBuddy Mini 2 offers 16-bit mono audio, whereas the BeatBuddy (whether version 1 or 2) are stereo 24-bit devices — 16-bit audio is ample for audio playback, of course (the same as used for CDs), but indicates the use of different components. But while there is a theoretical audio quality difference to consider, the Mini 2 does deliver very respectable sound quality, and if you don't need the more advanced functionality or the ability to import drum kits, then the Mini 2 may well meet your needs — for a fraction of the price.

■ The BeatBuddy Mini 2 cuts out what some users might find superfluous to give you a capable drum accompaniment pedal for a rather lower price.



browsers for Mac and Windows include Microsoft Edge, Opera and Vivaldi, so there are plenty of options.

Note that unlike some other products, there's no USB connectivity, so to use the editor software you must remove the SD card and place it in a card reader connected to your computer. BeatBuddy Manager can be used to create or edit a project, which can then be exported or sync'd to the SD card. The card is then replaced in the pedal — so while a direct USB connection might be desirable there's really not that much to it. BeatBuddy Manager includes a virtual BB2 pedal, which allows you to audition your projects before returning the card to the pedal. To avoid messing up the files on your SD card, you can copy the whole card to your computer and then work on the copy, before copying your work back onto the SD card.

There's a pleasing emphasis on 'bread and butter' rhythms in various genres that are uncomplicated enough to make them practical options in a number of songs. Those genres range from Latin and blues to reggae and soul, with a smattering of world music thrown in too. Rock, pop, ballads and country are all well-served too. But a new feature for BB2 specifically (but not for the Mini version) is Autopilot, which allows for the creation of what are essentially customised drum backing tracks for specific songs that still leave the user with the ability to interact with them in real time, should the arrangement need to be changed on the fly — for example triggering a fill, pause/unpause to extend a song section, or moving on to the next song part early.

On the included SD card you'll find a section of Autopilot example song structures based on a handful of well-known pop songs, and additional Autopilot songs are available for purchase on Singular Sound's Beat Boutique page. One of the songs included on the SD card is 'My Generation' by the Who, which is a song I know well and have played many times. Sure, it might not quite give you Keith Moon's vibe, but all the breaks and fills are in precisely the right place, and it works rather well.

Verdict

If you want to delve into every nook and cranny of the BeatBuddy 2, the options can get quite deep. But the beauty of the system is that you can use it right

out of the box. You just explore the additional features when you need them, and most of the time the pedal guides you through what you need to do anyway. Ideally, I'd have liked the ability to edit or create songs without having to take the SD card out of the pedal (so that the pedal could be left connected to a sound system to audition the changes), but the system works well enough as it is.

For most routine work, the included song rhythms and fills do a perfectly good job without the need to do much other than select a song and a tempo, so operation need not be complicated. Some of the beats may initially strike you as being a little 'straightforward', but for me that's actually a real

strength — the opposite of 'preset zero syndrome' (where presets are programmed to sound impressive rather than to be useful), this means they can be used to underpin a wide variety of songs without any complicated programming. Similarly, the fills/intros manage to do what's needed without getting over-fussy. The sounds themselves are solid and credible too. Well worth checking out. **///**

\$ BeatBuddy 2 \$599 (discounted to \$499 when going to press). Beatbuddy Mini 2 \$299 (discounted to \$149).

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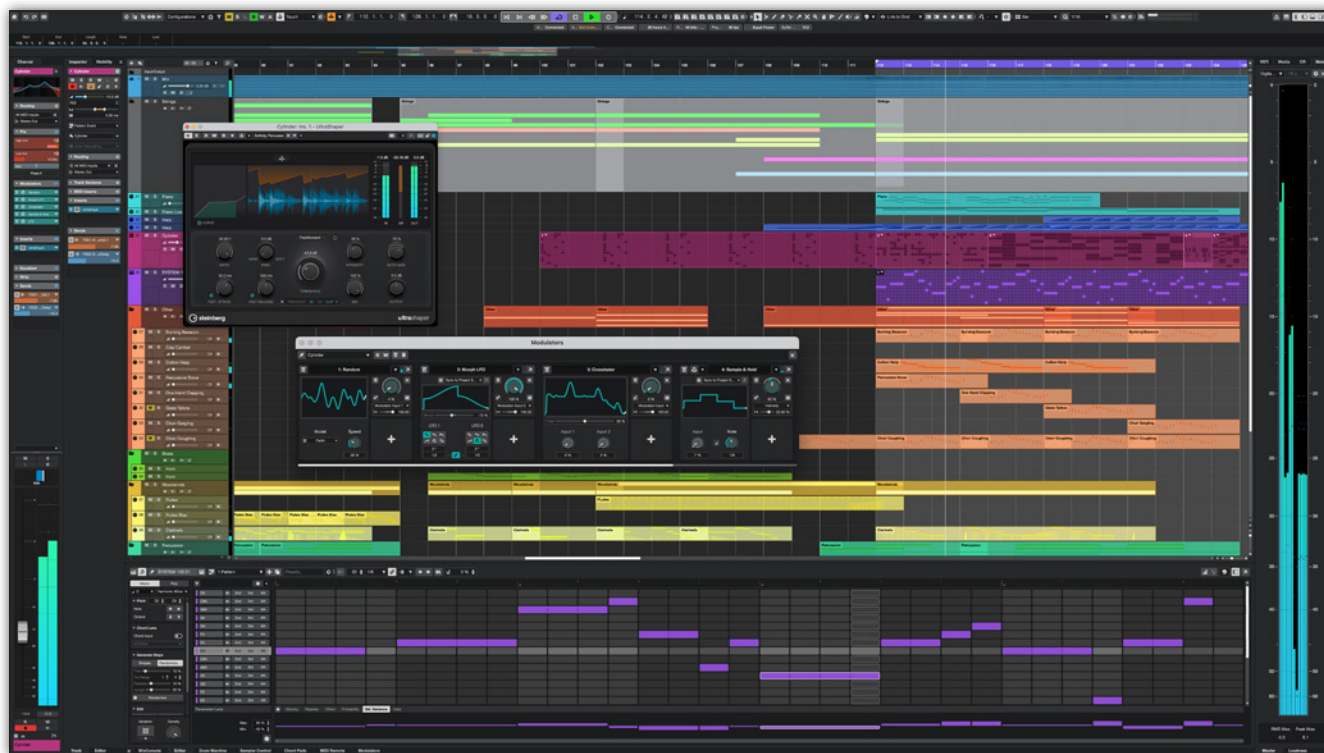
W www.singularsound.com

The advertisement for IsoAcoustics features a dark, moody background with a green-tinted overlay. At the top left is the IsoAcoustics logo, consisting of three concentric green arcs. To its right, the brand name "ISOACOUSTICS" is written in a large, white, sans-serif font, with the tagline "Sound quality by design" in a smaller font below it. In the center, the words "PRODUCT SELECTOR" are displayed in a large, green, sans-serif font. Below this, there are two circular images of speakers. The left speaker is a black, floor-standing model with multiple drivers, labeled "ISO-STANDS" below it. The right speaker is a smaller, black, desktop model, labeled "ISO-PUCKS" below it. Between the two speakers is a large QR code with the IsoAcoustics logo integrated into its center. Above the QR code, the text "SCAN TO FIND THE RIGHT AUDIO ISOLATION FOR YOU" is written in white. Below the QR code, the website "CALCULATOR.ISOACOUSTICS.COM" is listed in white. At the bottom of the advertisement, the phrase "CLARITY BEGINS WITH TRUE ISOLATION" is written in a large, white, sans-serif font, with "TRUE ISOLATION" in green. Below this, the text "OUR AWARD-WINNING AUDIO ISOLATION SOLUTIONS ARE THE EASIEST UPGRADE YOU'LL EVER MAKE" is written in a smaller, white, sans-serif font, with "EASIEST UPGRADE" in green.

Steinberg Cubase Pro 15

Digital Audio Workstation

What does the latest version of Cubase have in common with the 1982 science fiction movie *Tron*?



MARK WHERRY

Since the release of Cubase 14 last November, Steinberg have issued three maintenance releases to the company's "Advanced Music Production System". And while this wouldn't normally be notable — such updates have been common with previous versions — one of these releases contained far more than the usual handful of bug fixes. Version 14.0.20 might not sound like a striking step forward, but it brought substantive new features to what had already been an impressively ambitious upgrade. It also became the first official release to natively support ARM-on-Windows systems powered by Qualcomm's Snapdragon X Elite processors, such as Microsoft's latest Surface devices.

With so much momentum during the Cubase 14 era, an initial glance at the headlines for Cubase 15 might seem underwhelming by comparison; however, such an impression belies the work Steinberg have undertaken for this year's upgrade. Whilst some of the

new functionality inevitably builds on more recent additions to the application — particularly the Pattern Editor and modulators — it would be wrong to assume Steinberg's developers have focused on these areas exclusively.

Hub-A-Dub-Dub

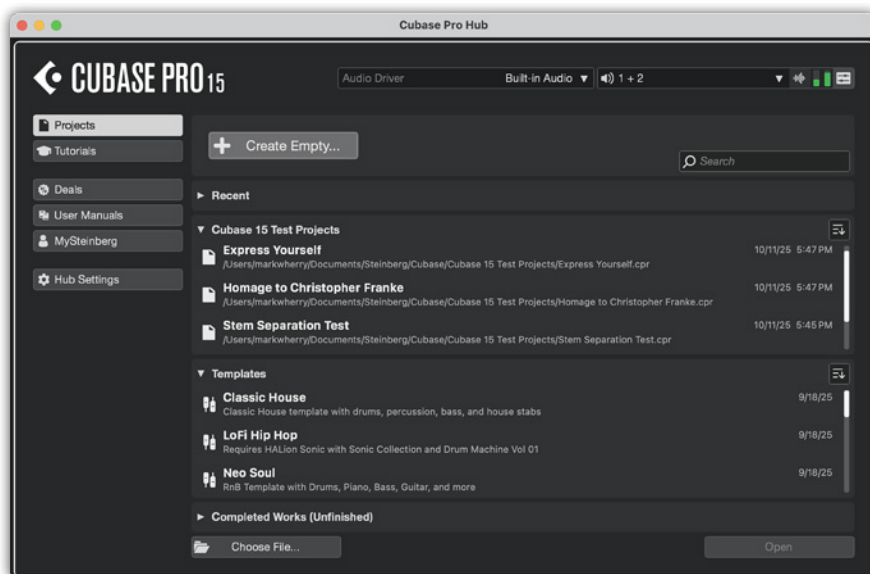
Since the release of version 7 over a decade ago, Cubase's Hub window has eagerly greeted users at launch, providing a starting point for creating new or opening existing projects. And while I can't say I'm a fan of such windows, since the amount of utility they provide rarely warrants their gatekeeping nature, Steinberg have finally rethought this feature in Cubase 15. The Hub's interface has been admirably overhauled, making it both more useful and more visually appealing.

A brief Guided Tour introduces the new Hub, reminiscent of a similar feature in Dorico. Although, unlike Dorico, the tour guide window jumps to each feature being described — instead of remaining in one place and highlighting the relevant feature

— and conveys too little information to be truly helpful for Cubase neophytes. Existing users will find it a brief nuisance, and although it only appears once, it will resurface if the Preferences are deleted or disabled, such as when running Cubase in Safe Mode.

The new Hub's centre area reorganises the recent projects and templates categories into sections, and additional sections can be included for custom locations via a new, dedicated Hub Settings window. This is a great way to provide quick access to Cubase projects belonging to different real-world projects, and, although you can't remove the Recent and Templates sections, it's possible to toggle the visibility of these sections and any others you might add.

Each section offers a 'sort' control to list projects by name or date in ascending or descending order. But the icing on the cake is a search filter to assist in finding projects across all sections in the centre area; the cherry would have been an option to automatically open and close



■ Cubase's Hub window has been redesigned with some useful improvements, such as the ability to configure the centre area with sections based on custom folders.

sections to reveal matching projects without having to do this manually.

Like the Hub in Steinberg's stage production application, VST Live, it's now possible to change the current audio driver and set the main stereo output directly from the Hub. Although, being somewhat lazy, it's perhaps a shame that — unlike Logic Pro's equivalent Choose a Project window — you can't specify a surround format for a new empty project in the Hub or activate Dolby Atmos, incorporating the behaviour of the Atmos Setup Assistant.

This is presumably because the Audio Connections window can manage input and output configurations independently of projects, which persist after a project is closed. So, if the previously open project used anything other than a stereo configuration for its main output, Cubase will substitute the Hub's stereo output menu for a button to open the Audio Connections window.

In addition to being able to set the audio driver from the Hub, Cubase 15 introduces a new Audio Settings window, consolidating the Audio System and Driver pages from the Studio Setup window into a single, dedicated panel. However, rather than serving as a replacement for these same settings in Studio Setup, the Audio Settings window duplicates this functionality, which seems like a missed opportunity to do something more radical.

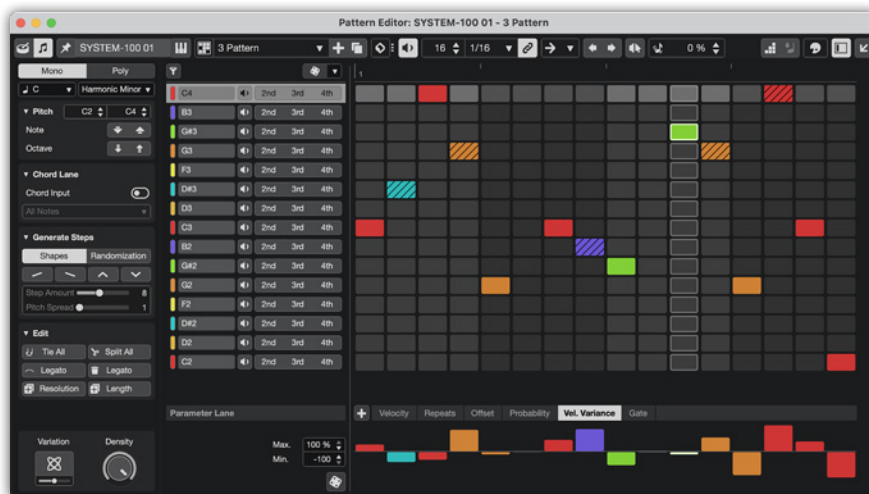
Behaviour Patterns

Although the Pattern Editor debuted alongside Cubase 14's Drum Track feature and was primarily intended for programming

drum patterns, its underlying nature as a step sequencer — triggering notes on different lanes — meant it could just as easily be used with instruments beyond drum machines. After all, step sequencers have long been used to craft TB-303-inspired bass lines or homages to Kraftwerk. The only catch was that such patterns are typically monophonic, whereas the Pattern Editor was conceived as a polyphonic sequencer for programming drums.

To address this minor inconvenience, Cubase 15 introduces a new Melodic Mode, providing the Pattern Editor with a complementary set of tools making it more versatile for both monophonic and polyphonic pitch-based sequencing.

Unlike Drum Mode — where lanes are derived from drum kit assignments and can be added or removed individually



■ The Pattern Editor's new Melodic Mode makes it easy to program sequences monophonically.

Steinberg Cubase Pro 15

\$579

PROS

- The Pattern Editor's new Melodic Mode makes it easier to program bass lines and similarly sequenced patterns.
- Six new modulators are more than the sum of their signals.
- A welcome reworking of Expression Maps, borrowing ideas from Dorico.

CONS

- Stem Separation isn't quite on a par with some implementations in competing products.
- Modulators still can't be used on MIDI tracks or easily routed from one channel to another.

SUMMARY

Cubase 15 builds on the functionality introduced in 14 — making features like the Pattern Editor and modulators even better for a broader range of musicians — without neglecting other areas of the application.

— Melodic Mode displays lanes for pitches within a defined range. All chromatic pitches within this range are shown on individual lanes by default, but you can also select a root note and scale so only pitches within that scale are shown. For example, setting the root note to D and the scale to Harmonic Minor displays lanes exclusively for pitches belonging to the D harmonic minor scale, which is — of course — the saddest of all keys.

These pitch constraints are particularly helpful when generating patterns automatically using Shapes or Randomisation. Shapes create rising and falling contours spread out over a number of steps, while Randomisation is carried over

»



■ Cubase 15 features six new modulators. Note the outputs of the Random and Attack Decay modulators are routed to the inputs of the Crossfader modulator. The Sample & Hold modulator is using the output of an audio track as a side-chain source.

» from Drum Mode and produces a sequence based on user-defined conditions.

Once a pattern has been generated or created manually, Melodic Mode allows further refinement via the Variation and Density controls. Variation transforms the pattern based on a specified amount of complexity, whereas Density non-destructively thins out the number of notes deployed. And, if a root note and scale have been set, changing the root note offers a quick and musical way to transpose the pattern.

Some enhancements to the Pattern Editor apply to both Melodic and Drum modes, such as the ability to edit parameters directly from the Step Editor. You can now click to add or edit a step, or keep the button held down and drag the cursor up or down to adjust that step's parameter value for the type currently selected in the Parameter Lane. This is

tremendously handy, although I still wish the height of lanes was adjustable.

Another welcome improvement is that it's now possible to move and copy Pattern Events between Drum, Sampler, Instrument, and MIDI tracks — albeit with some limitations. For example, Pattern Events can't be shared with tracks if they already contain Events created using a different Pattern mode, which basically means you can't mix and match Pattern Events created with Drum Mode and Melodic Mode on the same track.

Modulators For The Users!

Modulators were another great addition in Cubase 14, and Steinberg have also been busy improving this feature in version 15 with the inclusion of six new modulators. These offer even more ways to generate signals and, perhaps more importantly, the ability to combine modulators into composite sources.

The aptly named Random modulator includes models inspired by the work of a computer scientist, a meteorologist and a biochemist. And while these occupations sound like the setup for a particularly nerdy pub joke, Steinberg's developers had good reason to delve into such research: generating random numbers isn't as straightforward as you might think. And generating musically useful random signals is even harder. Interestingly, one

of the models is named after Ken Perlin's eponymous noise algorithm, which was originally developed for the movie *Tron* as a way of making surface textures in computer-generated imagery look more natural.

The 'wave' in the Wavefold LFO modulator refers to a single oscillator capable of generating a sine or triangle wave — or anything in between via a Shape control. Folding describes how the oscillator's output is manipulated such that, if the value of the signal exceeds a specified threshold, the waveform is inverted — or 'folded' — back on itself. And of course, like any modulator control, the controls for Shape and Fold can be modulated, which only adds to the fun.

Attack Decay is an imaginatively named envelope generator, although a more suitable name might have been Impulse Decay since only the time and shape of the decay curve is adjustable — the attack stage is fixed with 100 percent amplitude and zero length. And while you could essentially create the same result more flexibly with the existing Shaper modulator, Attack Decay is simpler and allows the shape of the curve to be easily modulated.

The new Sample & Hold modulator works as you'd expect: the current value of the input signal is 'sampled' and then

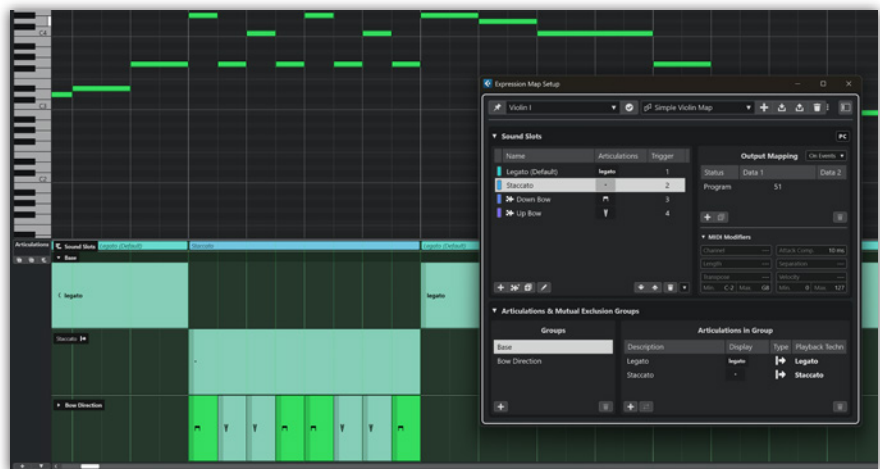
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Scaling New Heights

In addition to supporting high-DPI modes, recent Windows versions of Cubase have also offered an Application Scaling setting, allowing the user interface — and that of well-behaved VST3 plug-ins — to be scaled independently of the system's setting. In version 15, Steinberg not only bring this feature to the Mac, but enable changes to take immediate effect (on both platforms) without the ceremonial restarting of Cubase demanded by the earlier Windows implementation.

This newfound rescaling ability on the Mac will be welcomed by users with Retina displays, especially those lacking an option between More Space and their native resolutions — which appear to have been designed for birds of prey. Reducing Cubase's interface by, say, 25 percent, allows more of the application to fit on-screen without affecting the rest of the system — and this feature is even handier now that Cubase 15 supports macOS's dedicated full-screen mode.

Curiously, while the Application Scaling setting now sensibly resides on the User Interface preferences page, the Enable HiDPI control in the Windows version has been left marooned under General. Given that these two settings are closely related, it seems odd they wouldn't be kept together — as indeed they were in version 14.



■ Both the Expression Map Setup window and the way in which playing techniques are represented in the Key Editor's Articulation Lane have been redesigned in Cubase 15. Note the 'Bowing Direction' group has been consolidated into a single lane.



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» 'held' at that value for the specified time, defined in either note length or Hertz. And finally, the Morph LFO and Crossfader modulators are similar in that both enable you to mix — which is to say 'morph' or 'crossfade' — two signals into a single modulation output. The difference is that Morph LFO provides two built-in oscillators, whereas Crossfader features two input knobs that can be modulated by the output of other modulators.

Unlike the improvements made to the Pattern Editor since its introduction in Cubase 14, many of the modulator shortcomings remain unaddressed. For example, modulators still can't be used on MIDI tracks, and only destinations available on the same track/channel as the modulators themselves can be modulated directly. On the plus side, though, modulating a track's mute control now works, except I did notice a visual anomaly where a modulated mute control can stop indicating its state on the relevant Track Header.

Cubase 15 also aims to unify the behaviour of the various bypass controls used by modulators. Whether you're bypassing modulators at the connection, modulator or track level, any connected destination should, in theory, be restored to its original, root value when bypassed. However, in practice, quirks in how the root value is interpreted can induce some head-scratching moments.

For example, if you add an LFO modulator and connect the track's pan control as a destination, bypassing that connection sets pan to the value assigned

by the connection dial. If the dial is centred, the pan neatly reverts to the centre position when bypassed, which seems fairly straightforward. However, if you establish the same connection using the new Wavefold LFO modulator, for instance, the bypass behaviour is slightly different.

Although bypassing at the connection level will again restore the pan control to the position of the connection dial as expected, bypassing the Wavefold LFO modulator itself — or the entire track on which it resides — sets pan to a level

Separation Anxiety

Perhaps one of the least interesting new features in Cubase 15 — and to be fair, its intrigue is only diminished due to such functionality becoming *de rigueur* in the current generation of competing products — is the integrated ability to separate stems from fully rendered audio files.

Steinberg's Stem Separation can unmix up to four stems from a selected audio event — vocals, drums, bass and other — which is similar to other implementations. Dedicated products like Rip-X offer considerably more advanced capabilities, of course, though a more *apropos* comparison is that Logic Pro's equivalent Stem Splitter can extrapolate six stems in the current version, keeping piano and guitar separate from the 'other' stem.

Stem Separation in Cubase is configured from a dedicated window, which can be opened by choosing Separate Stems from the Audio menu or by assigning a key command — sadly, you can't summon it from the Project window's context menu. This window lets you toggle which stems to separate; and, if you select only two or three of these, Cubase can optionally combine the selected stems into a new audio file rather than creating individual stems.

The results will vary depending on the source material. But, listening back to a mix of the stems summed at unity gain, the sound was noticeably more band-limited than the original file, with the individual stems containing a little more leakage than I've heard with other algorithms. Although it's worth noting such analysis is subjective and quantifying the integrity of the stems generated is surprisingly difficult.

While it's perhaps tempting to consider comparing the phase-cancelled results from such algorithms with the original, this kind of assessment reveals very little about any qualitative differences. Many separation processes enforce a degree of mixture consistency to provide more musically useful results when recombining stems, which is why you can re-sum all the stems except the vocal one, as an example, without too many unwanted artefacts.

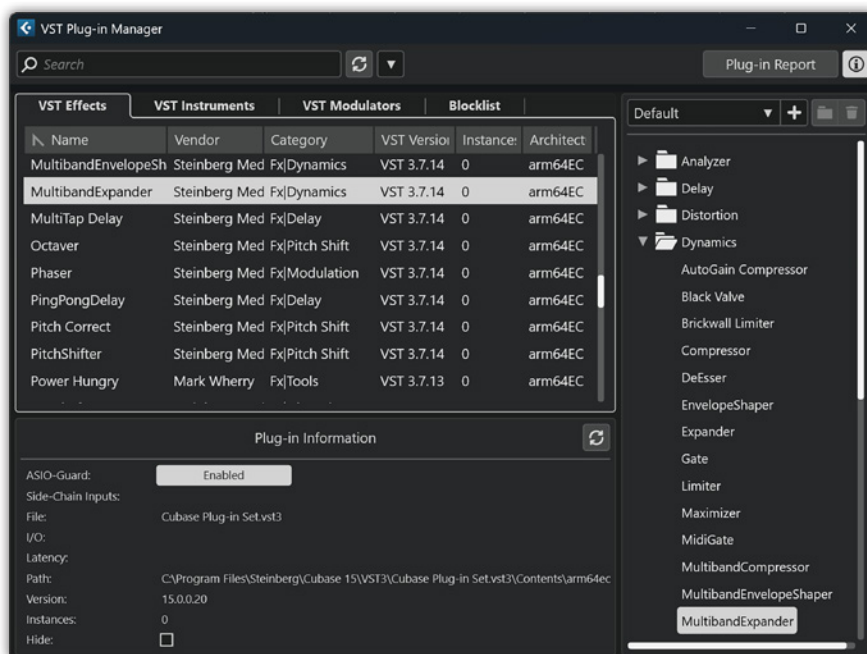
Stem Separation wasn't functional for the initial Windows-on-ARM version of Cubase 15, although it will be enabled in the 15.0.5 update that should be available by the time you read this.

that's the sum of the dial's position and the specified modulation depth. It's somewhat confusing, leaving you to wonder if this is a bug or a feature.

X Marks The Expression

One area of Cubase that's remained largely unchanged since its introduction back in Cubase 5 in 2009 is the ability to work with articulations and playing techniques — especially when writing for orchestral instruments — using Expression Maps. Indeed, the ideas behind this feature have arguably evolved further since Dorico introduced its own implementation. And the reasons are perhaps obvious: whereas playing techniques are fundamental in notated composition, production-oriented applications like Cubase can tolerate a more *laissez-faire* attitude.

However, as sample libraries have become larger and more complex, the use of Expression Maps in Cubase has only increased, which is presumably why Steinberg finally chose to revisit this feature in Cubase 15. And since the interoperability between Cubase and Dorico has been steadily improving over the years in both concept and compatibility, it makes



■ The Plug-in Manager's appearance has been modernised in Cubase 15. Because this screenshot was captured on the Windows-on-ARM version of Cubase, the Architecture column identifies plug-ins that natively support the ARM64EC environment.

sense Cubase's developers would turn to Dorico for inspiration.

In Dorico, articulations are triggered by switches, of which the two main types are base switches and add-on switches. Base switches define core articulations such as legato, staccato or tremolo, while add-ons describe complementary techniques that can be added on, such as up- and down-bow directions for a violin.

For Cubase's Expression Maps, Sound Slots serve a similar role to switches, and in earlier versions a slot could comprise up to four articulations with each being placed in one of four numbered groups that operated exclusively. This meant if your base articulations were placed in the first group of each slot, only one would be active simultaneously, although more complicated mappings could quickly become confusing.

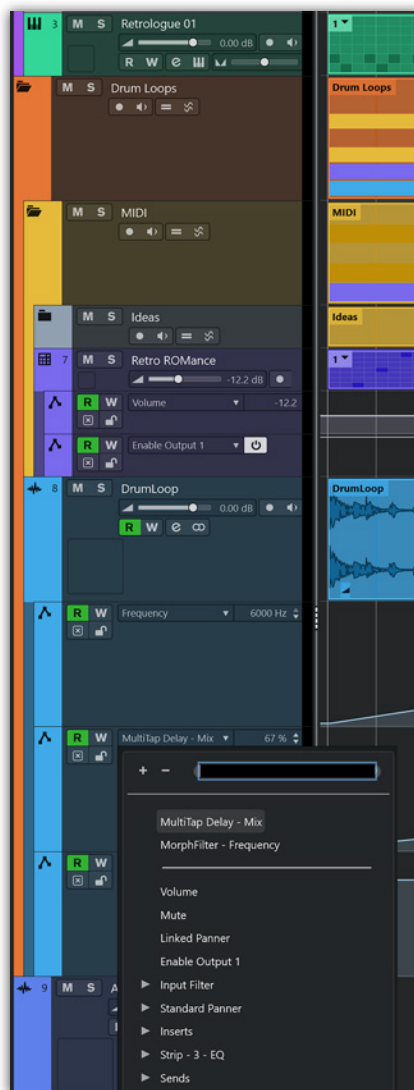
In Cubase 15, these four, numbered groups have been replaced by an unlimited number of named groups for each map, accompanied by Dorico-inspired Add-On Sound Slots. These changes affect the way Articulation Lanes are used and displayed in the Key Editor, offering a clearer organisation, with the ability to collapse groups into single lanes. And the Articulation Lane is complemented by another Dorico-inspired feature, which optionally displays a Sound Slots lane in the Key Editor to indicate which slot is active at any point in the timeline.

Given that Dorico could already import Cubase's Expression Maps, it makes sense that Steinberg would add support to import Dorico's Expression Maps in Cubase 15 — which can be dragged and dropped onto the updated Expression Map setup window just like any other Expression Map file. However, the import is quite limited and currently restricted to slots within a single group, which seems a little ironic: despite Cubase borrowing Dorico-style add-ons, it can't yet import them.

The only minor grievance I had with the modernised Expression Map setup window is that it no longer manifests itself as a floating tool window, meaning that it doesn't stay on top of other open windows. And even though a customisable toolbar has been added, there's no Always on Top option (as is the case with many such windows in Cubase) to restore the prior behaviour.

Odds & Ends

As is always the case with a major Cubase upgrade, there's a long list of smaller improvements that could easily double the



■ Folders and automation tracks now use indentation colour to make ownership clearer. Notice also the new automation parameter menu.

length of this review if I mentioned them all. However, a few worth noting include the ability to add volume and pan controls to track headers. These resemble the horizontal volume and pan controls used in the Inspector and are particularly beneficial when you want to maximise screen space by hiding everything except the track list and event area.

Another change related to track headers concerns folder tracks. Previously, a folder's colour was indicated by its colour bar — or the entire track header if Colourise Tracks is enabled — and any tracks within a folder were indented with black space. In Cubase 15, a folder's colour is now used as the indentation colour, making the relationship between a folder and its contents much clearer — assuming you've chosen your colours sensibly, of course!

Unlike previous versions, assigning a colour to a folder track in Cubase 15 causes all tracks within that folder — and any tracks added subsequently — to inherit the same hue, which is a move that might not necessarily be hailed as progress amongst longtime users. Happily, a new option in the Event Display (Folders preferences page, Apply Colour of Folder Track Automatically) allows the previous behaviour to be restored by setting it to Never rather than Always.

This new indentation style extends to automation tracks, where the inset now appears as a darker shade of the track's colour, making the allegiance of automation tracks more obvious. There are some neat improvements for quicker access to automation parameters, with the automation pop-up menu now divided into two menus: one for parameters — which is searchable and displays the five most recently used — and another for commands. The new parameter menu is such a useful addition, I wish somebody had thought of it sooner! (I also wish Drum Machine's automation parameters were less obfuscated, but you can't have everything.)

Finally, given how much functionality has been added to the Export Audio Mixdown window over the years, Cubase 15 introduces a handy Quick Audio Export command for expediently exporting a stereo mix. The export range is automatically determined based on event locations (including reverb tails), so the only decision you need to make is whether to save the rendered output as a 24-bit WAV file (at the project's sampling rate) or an MP3 file (encoded at 256kbps with a 44.1kHz sample rate). And while it would be easy to criticise this feature for omitting 'insert-any-option-here', such criticisms would rather miss the point!

Fifteen To One

In many ways, Cubase 15 feels like the conclusion to what would have made Cubase 14 the perfect release. And while such a remark runs the risk of sounding like a back-handed compliment, it very much isn't. Cubase 15 is far more than merely a polished iteration of the prior release — version 14.0.20 already delivered that — since it enhances what came before whilst still adding multifarious new functions. ■■■

\$ Cubase Pro 15 \$579.99, Artist 15 \$329.99, Elements 15 \$99. Upgrade pricing available.
W www.steinberg.net

MODULAR

Rides In The Storm SED-CSM

Eurorack Module

You can understand my scepticism upon learning of the SED-CSM from Berlin's Rides In The Storm. A full analogue synth voice, with its own mixer, packed into a big-but-really-very-small 36HP, it's the kind of module one might consider for their very first Eurorack purchase, sitting comfortably alongside the likes of Intellijel's Atlantix, Verbos Electronics' Sawtooth Stack or Black Corporation's ISE-NIN in terms of functionality. Only, those three retail at around \$699, \$1069 and \$999 respectively. The SED-CSM? A tidy \$649, or thereabouts. Too good to be true?

It's worth me saying that, if you aren't familiar with RITS as a developer, the output of Uwe George Giegler's company strikes me as somehow very impulsive, in the most respectable way. I dare say there's something very

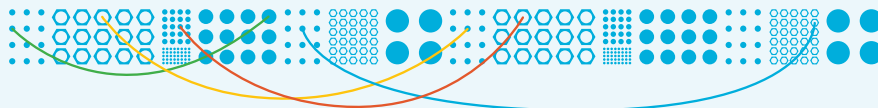
punk at the heart of RITS output. It's this level of impulsiveness that I'm sure led to the SED-CSM. It has controls and functions flung onto it with abandon, a manual which amounts to little more than a diagram of what each control is (it's written on the panel already) and with scanned-in handwritten annotations on setting the rear jumper switches. Fantastic. But immediately upon mounting it into my rack I could see that Giegler isn't messing around. For a start, the build quality is lovely. The panel is chockablock with functions to the point of feeling quite congested, but fair to say everything here still manages to feel solidly mounted and firm, with all knobs, sliders and buttons comprising quality tactile components. And besides, a panel feeling like it has 'too many functions' is, in the scheme of things, not a bad problem to have.

As for those constituents of the SED-CSM's panel themselves, many are usual suspects when it comes to a subtractive synth voice, but there are more than a few curveballs here to separate it from the crowd — most notably, in my view, when it comes to movement. The two

oscillators each offer pulse, saw and triangle waves, with wavefolding on offer for VCO 2's triangle. There's also generous capacity for sub oscillation with variable waveforms and XOR ring modulation available, attenuated via the cute onboard mixer, along with white noise. There are two ADSR envelopes, two LFOs and a 24dB/octave low-pass filter. Of course, one expects CV inputs for a raft of functions on a module like this, and I wasn't disappointed here — particularly since each oscillator has its own jack for 1V/octave pitching and a Reset input, and each envelope has its own trigger outputs to mark its attack and decay stages.

It follows that re-routing various signals about the place with internal patching is always on the cards, but of course with a full synth voice like this, internal, patchless functionality is just as important. There's a generous slab of harmonic texture on offer right away here, but the SED-CSM goes further with things like variable modulation of VCO 2's wavefolding — either manually, with an LFO or with an envelope — and the capacity to modulate both oscillators' frequencies independently with each of the two envelopes, both of them together with LFO 2, or a combination of these. There's a Sync button, variable pulse width for each oscillator, and also a Gravity button to change the behaviour of the sub-oscillators (as far as I can tell, by dropping its octave, but it sounds like there's also some sync going on — as you may imagine, this isn't made clear in the manual). I'm always happy to see the trans-coast pairing of wavefolding and filtering, particularly a filter that can be driven, as is the case here. While the SED-CSM's filter is certainly on the characterful side when it comes to analogue variability and squealing self-oscillation, it's nonetheless substantial and smooth, and the Drive button is a welcome addition for





squeezing more tone out of it if needed. The combination of wavefolding, drivable filtering, a ring modulator and noise makes for a fantastically aggressive synth voice if that's what you desire.

There's also something of a secret oscillation weapon up the sleeve of the SED-CSM, which is that it's possible to engage 1V/octave tracking on both of its LFOs. This might look like expressive vibrato that moves in response to the note played, or — in the case of LFO 2 — audio-rate modulation that becomes a *de facto* third oscillator for some Minimoog-style massiveness. LFO 2 can switch between (shape-variable) triangle, pulse or random voltages,

meaning that it can take on various tonal characters of its own or indeed assume noise duties alongside the conventional noise generator.

In addition is a manual envelope trigger button — always handy for quick referencing when sound-sculpting — alongside another West Coast-inspired option, which is for either or both envelopes to cycle. This can of course mean that they take on LFO-like roles; so pulling in an external envelope to take care of conventional functions, I could access a total of four asynchronous LFOs all churning away together, which I'd consider generous on a standalone synth of double the SED-CSM's size!

Eurorack has long had a reputation for being untenably expensive for many, but every now and again a developer comes along with a design that bucks that trend, presenting buckets of character that won't break the bank. Its idiosyncrasies might take a little getting used to for some, but the SED-CSM is an excellent example of this, suited both to existing users looking for a highly flexible standalone voice, or to newcomers looking for a vital point of access that will perform marvellously by itself, right out of the box.

William Stokes

\$ \$649

W www.rides-in-the-storm.de

Strymon SuperKar+

Eurorack Module

You would be forgiven for hoping that Strymon were bringing Supermarionation to Eurorack with some kind of strung-up Supercar integration. However, this is SuperKar with a K and the all-important '+' that puts the strings firmly inside the module rather than dangling from them. This module is based on the Karplus-Strong algorithm that uses short delay circuits to produce surprisingly lifelike plucked string and

vibrating pipe tones. It's 'Super' because the SuperKar+ can produce up to 32 voices and has a bi-timbral split to offer a Chord voice and a separate polyphonic Solo voice all within the one 12HP module.

The design is dark and brooding, with imposing knobs and a pair of intently staring green LEDs. The labelling is very clear with a trace of a dotted line separating the Solo and Chord voice patchbays and wandering off to delineate between the controls. It's a very handsome module. However, it quickly becomes apparent that the knobs are a little too imposing for the space and access to every control is not very comfortable.

Once you start playing with the SuperKar+, most of the design frustrations melt away in response to the sheer beauty of the sound. Most modular users will be familiar with the sound, probably via Mutable Instruments' Rings, or in my case via the 2HP Pluck and Qu-Bit Surface. What I find most interesting about the SuperKar+ in comparison is how restrained it is. The 2HP Pluck will often squeak at the higher end of things, and the

Surface could easily drown itself in sustain, whereas the SuperKar+ is thoroughly well behaved throughout.

The Solo voice is on the left and you have three main controls: Damp, Decay and Attack. Damp refers to dampening of the strings, pulling up from heavily muted to ringing free, whereas Decay governs how much space is given to the ringing. You'll find there's a lot of interesting interplay between the two. Attack is not a rising envelope attack; it's more of a move from pluck to bowing action, and it doesn't really go much past staccato. It's oddly authentic and sounds great in most cases, and a bit weird in others, depending on what you're doing. With the second pipe-like timbre, the Attack is used more to introduce breath and overblowing, which definitely brings a convincing bit of breathy woodwind to the module.

You also have secondary controls — accessed by holding the right Timbre button — over level, glide and polyphony amount. The knobs do that annoying thing where once you've moved a secondary parameter, the primary parameter will jump to where the knob is now pointing as soon as you touch it. Some kind of soft takeover

would be appreciated. When using CV, the polyphony is only relevant to the sustain of notes continuing to play as more come in. However, plug in a MIDI TRS cable and you have up to 16 voices of SuperKar+ happiness at your fingertips. The MIDI implementation is minimal; you have notes and velocity, that's it, and it's only for Solo mode, not Chords.

On the right, we have the Chord voice which gains an additional Harmony knob. There are 15 harmony options ranging from the root note, and then dialling up it adds fifths and octaves in various combinations. The right side of the knob adds 'smart harmonies', which include thirds, sevenths and ninths that all respond to the Scale setting that you can choose via the knob's secondary function. So, as you feed the Chord mode a root note, you can use the knob or CV to reveal and remove extra notes. The scales are not CV controllable, which leaves you with all the chords having the same quality unless you change them manually. So, for instance, you can't sequence in a change from major to minor, which might be something you'd do in a chord progression.

Solo and Chord mode were designed to play together, and



» it's quite thrilling to be able to feed the Chord voice some root notes and then play the Solo voice over the top. Both voices end up at the same outputs, but we have some useful options here. You can set the outputs to give a wide stereo feel with alternate notes from the Solo voice being panned hard left and

right, or pull them closer to the centre with the Chord voice in stereo. There's a mono mix option, or a split option where the left carries the Solo and the right carries the Chord, which is great for processing them differently.

Both sides offer an audio input so you can use the Karplus algorithm as

a sympathetic resonator array. This opens the SuperKar+ up to all sorts of tonal possibilities.

The SuperKar+ is a well-thought-out module with a stunning sound and a habit of occupying your time. The Chord and Solo voices give it multiple layers of usefulness in your rack with plenty of tonal

variations to explore and modulate. It would benefit from greater MIDI control, CV-controllable scales and slightly slimmer knobs, but otherwise it's a superb implementation of Karplus-Strong, and a lovely piece of work. *Robin Vincent*

\$ \$399

W www.strymon.net

Modular Profile: Uwe George Giegler

Founder of Rides In The Storm, Berlin-based Uwe George Giegler is a designer whose precocity is matched only by his talent at circuit building. Having rubbed shoulders with some of Germany's most revered synth developers throughout his career, Giegler continues apace to create modules that are not only original in design and expertly engineered, but also affordably priced.

On his entry into modular

Back in the early '80s, I was already designing concepts for modular systems and working with various circuits. Later, I saw Dieter Doepfer's first modular releases at the Frankfurt Music Fair, where I also met Andreas Schneider. After that, Doepfer modular walls could be found in many other studios in Frankfurt! A little later, I began working with Manfred Fricke, MFB, and we continued to work together until his death. I then founded my own company in 2017 with Frank Kumm: IO Instruments. When Manfred became very ill and then passed away, I founded a second company of my own, Rides In The Storm. I was able to recruit circuit designer Nikolai Dichev, who previously worked for ACL, and later Moritz Goetz, who develops software.

On his go-to modules

Oh, I'm very spontaneous! A lot of modules, especially from small manufacturers, have an idea that I like to look at. My focus is on sound and originality; I like to keep track of things, otherwise I lose interest. I remember hearing a great module — with mini tubes — from a US manufacturer, and I was instantly composing music in my head.

On the story of Rides In The Storm

The name came to me after a visit to a bar. I was still a little tipsy, and 'Rides In The

Storm' popped into my foggy brain. There are associations with the Doors and the melodious sound of ride cymbals in the wind, which is the actual meaning. First, I created a sound page with all kinds of patterns and sounds for synths and drum machines. But it quickly became clear that I wanted to make my own series of synth modules, with excellent sound quality at affordable prices in a compact footprint.

Little by little, over 25 products have been created to date. Together with Nikolai, we swear by discrete circuits and our own developments, but that also means that the development time is quite long and we need quite a few prototypes before it comes onto the market. Nikolai now lives in Bulgaria, so we visit each other regularly for work meetings. This also brings a certain slow-down, which I consider urgently necessary in this field. Our next release will be the QSQ sequencer and for next year we're working on a standalone drum machine, Rhythm Rides.

On the SED-CSM

The SED-CSM is killer synth! The sonic possibilities are enormous; the transients, the punch, the definition is great... Four years of development, as well as the necessary perseverance through highs and lows, preceded it. All our experience and sonic ideas have gone into this instrument. We are very proud of it. The synth is versatile, the tuning is stable — thanks to THAT 310 Transistor Arrays — it's discrete, powerful, compact and affordable. The idea originally came from an employee at Perfect Circuit in Burbank during one of my visits: 'Why not build a voice synth?' Well, here we go!

Another great thing about the module is that it can be connected to a second filter, either in series or in parallel; ie. before the VCA and after the internal VCF. Which is really fun.



There was no model for the SED-CSM; its sound is unique. There weren't many discrete circuits; most synths work with dedicated chips, like the Curtis. There is one exception here, though: the LFOs are made with 3345 chips, which also work well at audio rate and can be used as additional VCOs.

On the culture of modular

I can't really presume to talk about it, because there's everything in there — including the zeitgeist of individualism. But there are also jams here and there, and some people are good at it. It's a kind of hipster rebirth of the hippie era! In my workshops, I always ask everyone personally what their approach is. There are many answers: playing, contemplation, curiosity, musical interest...

Personally, I like to use modular as an addition in the studio, to make samples and loops, and I still use a groovebox live. But I also get up at night sometimes to try things out. Maybe it means that childhood can be extended, and in fact modular represents eternal life, we just don't know! *William Stokes*

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SAM INGLIS

Most of the products we review in *Sound On Sound* are good. Companies who make bad products don't tend to stay in business for long! Even so, it's rare to encounter something that instantly screams "modern classic", but Austrian Audio's first product was one such. Released in 2019, the OC818 is a multi-pattern capacitor microphone based around a re-engineered take on AKG's celebrated CK12 capsule. Employing a ceramic surround rather than the original "brass ring", Austrian Audio's CKR-12 is more consistent, more stable and easier to manufacture than the original — but, crucially, it shares all of its positive sonic qualities.

Had Austrian Audio simply used this capsule in a faithful recreation of the C414, it would no doubt have been very popular. But what makes the OC818 a modern classic is that they opted not to be limited by AKG's 50-year-old design choices. The OC818 dispenses with the C414EB's output transformer, improving its specs and producing a cleaner sound. It uses microprocessor control to adjust the polarisation voltages to each side of the

Austrian Audio

OC-S10 Capacitor Microphone

The OC-S10 is a premium vehicle for Austrian Audio's highly regarded CKR-12 capsule.

capsule, for superb polar pattern accuracy. Cutting-edge manufacturing techniques deliver unprecedented consistency, meaning that any two OC818s can be used as a pair with no need for additional matching. And not only can the polar pattern be controlled remotely over Bluetooth, but the signals from the front and rear of the capsule can be recorded independently, whereupon Austrian's Polar Designer and Stereo Designer plug-ins enable far greater control over the directionality of the mic than would be possible with switches alone.

What's perhaps most impressive of all about the OC818 is that it remains relatively affordable, retailing at well under £1000. That's not something

you can say of many modern classics, or indeed of second-hand C414EBs. What's more, Austrian Audio have used the same capsule in two even more wallet-friendly mics, the fixed-cardioid OC18 and OC16. They've also diversified into headphones, pencil mics and stage dynamic mics, with solid results. Until now, however, they haven't entered the premium tier of the mic market.

Launched at this year's NAMM Show, the OC-S10 is very definitely a premium studio microphone. In creating it, Austrian Audio haven't steered away from the trusty CKR-12, but nor have they taken some of the more obvious paths open to them. They could, for example, have used this capsule as the basis for a high-end

Austrian Audio OC-S10

\$2992

PROS

- Gorgeous, natural, open sound.
- Superb shockmount and pop shield design.
- Dual-output configuration makes it very versatile.

CONS

- You can buy two OC818s for less — and those are great mics too.

SUMMARY

Austrian Audio have pulled out all the stops to create their new flagship model, and the results are most impressive.

valve mic, or added an expensive output transformer and fancy Japanese capacitors. They could even have made a stereo mic. What they've actually done, however, is more subtle.

Doing The Rounds

The OC818, 18 and 16 all share a distinctive body shape, which is a little reminiscent of a hip flask. In its way, this is as individual and recognisable as the angular C414 body, and has some advantages over that design, including reduced susceptibility to RF interference. The OC-S10, though, looks nothing like either. Its generously sized, cylindrical body is outwardly more conventional than that of the OC818, but in its own way represents an equally innovative design.

Five basic polar patterns are available — omni, subcardioid, cardioid, hypercardioid and figure-8 — and the OC-S10 has the same complement of pad and high-pass filter settings as the OC818. Rather than being adjusted using switches, however, these are selected using detented rings that surround the entire body of the mic. These are satisfyingly solid in operation, but more practically, they make the current setting easily visible from any angle, since the legends for each position are reproduced on the rear side.

The OC818 already has an unusually open headbasket, but the OC-S10 goes even further, suspending the capsule in a considerable volume of free space. A similar three-point elastic capsule mount is used, and this in itself offers a decent degree of shock protection, but one of the OC-S10's outstanding features is the dedicated shockmount that Austrian Audio have developed. A threaded ring holds the mic in place

within an inner cradle, which in turn is suspended within a rigid outer frame. The linkages between the two are not formed from conventional elastic bands, but from Y-shaped pieces of soft rubber. These provide support in compression as well as in tension, and so even though the mic itself is only attached at one end, there's minimal droop when used horizontally.

This mounting system is remarkably good at preventing the transmission of mechanical noise. In fact, I'd go as far as to say that this is the most effective shockmount I've ever tested, especially considering the not inconsiderable size and weight of the mic itself. Tap your finger on the mic stand or even on the outer frame of the mount, and you'll hear nothing but what the mic is picking up through the air — no rumbles, thumps or extra low-frequency components at all.

My favourite feature, though, is the universal ball-and-socket joint through which it attaches to the mic stand. This permits a wide range of rotation in three dimensions, and holds the mic securely at whatever angle you choose. Similar mechanisms are offered by specialist manufacturers such as Triad-Orbit, and until you've tried them, you won't realise what a difference they make to the usability of a mic.

In addition, the OC-S10 comes with its own custom pop shield. This attaches magnetically to the mount, and disperses wind blasts using a fairly thick but lightweight screen made of foamed metal. In practice, I found that the dual-layer mesh in the mic itself was already reasonably effective at minimising pops, and the pop screen completely eliminates them. I'm never 100 percent convinced that pop screens are sonically transparent in other respects, but this one is as good as I've heard, with no ringing and minimal high-frequency loss.

Splitting The Difference

As previously mentioned, the OC818 offers remote pattern control courtesy of the optional

OCR8 Bluetooth dongle, plus the option to record the output from the rear side of the capsule separately, enabling the pattern to be decided after the fact. This is the one aspect of the OC818 design that sometimes feels a bit clumsy, because the second output appears on a mini-XLR socket, and the necessary adaptor cable tends to foul the mic clip.

Bluetooth control is not supported on the OC-S10, but dual-output recording very much is, and its larger body has enough room for two full-sized XLRs. These are positioned side by side, at the base of the body where nothing can get in the way. The obvious risk with this idea is that in normal use, when you're only connecting one cable, you might accidentally plug it into the wrong socket. Austrian Audio have prevented this through an ingenious mechanical locking arrangement, which blocks access to the second XLR unless a cable is plugged into the main socket. »



■ The foamed metal pop shield attaches magnetically to the shockmount.



► Polar pattern, pad and filter settings are chosen using indented rings that rotate around the body of the mic.

» Finally, premium mics are often distinguished by premium packaging, and the OC-S10 is no exception. The mic itself comes in a very nice wooden box with a magnetic closer, which is almost identical to the boxes Extinct Audio use for their BM9s. This, in turn, is housed in a robust hard-shelled case with compartments for the shockmount and pop filter. Austrian Audio put each production mic through a rigorous test procedure, and individual frequency response charts are supplied for each of the five polar patterns.

Whereas the OC818 has a thoroughly modern look and feel, the OC-S10 package embodies a slightly more timeless approach to industrial design. It's not retro, but nor does it wear its innovations on its sleeve in quite the same way. The overriding impression is of a seriously high-class product, constructed to the highest standards without any compromises. I think Austrian Audio have struck a really nice balance between form and function here.

Twin Peaks

The OC-S10 requires standard 48V phantom power to operate, and draws a modest 3mA or less. Its specifications are, unsurprisingly, very similar to those of the OC818, although it slightly betters that mic in a couple of ways. Self-noise is just 8dBA, a 1dB improvement over the 818, and sensitivity is 14mV/Pa to the 818's 13. On the down side, there's 1dB

less headroom, but since the maximum SPL is still 157dB, that's unlikely to be a problem in the real world. Output impedance is considerably lower, at 108Ω to the OC818's 275Ω.

As previously mentioned, the pad and filter implementations are the same as those of the 818. The pad operates in two 10dB stages, the first of which reduces the polarisation voltage of the capsule, while the second introduces attenuation in the internal circuitry. The corner frequency of the filter can be set to either 40Hz or 80Hz, with a 12dB/octave slope, or to 160Hz, in which case the slope is 6dB/octave above 80Hz and 12dB/octave thereafter. This offers a good array of options for tackling both subsonic noise and low-frequency build-up due to proximity effect.

The individual frequency response charts for the review mic corresponded closely with the generic measurements published in the manual. These, in turn, display a fairly strong family resemblance with the equivalent charts for the OC818 — and, completing the chain, with many mics based on the original AKG CK12 capsule. Mics based on this capsule design tend to exhibit the same general character, with a flat or even recessed midrange and a generous high-frequency lift. This treble emphasis often appears as two separate peaks on the response chart, which assume different proportions depending on the polar pattern.

In the OC-S10, the upper peak is the broader. Centred around 10kHz and spanning nearly an octave, it's at its most obvious in omnidirectional mode, being almost

5dB up. As you shift the pattern towards cardioid, this starts to flatten out and a second peak starts to develop, centred around 5kHz. By the time you reach figure-8, the 10kHz peak is barely 2dB

above the nominal zero, whilst 5kHz is boosted by as much as 5 or 6 dB.

In practice, although some shift in on-axis tonality is audible as you change

the polar pattern, it's more consistent and less obnoxious than it is in many large-diaphragm mics. I tend to avoid using my U87s and C414EBs in figure-8, as that upper-midrange emphasis can make things sound edgy, but the OC-S10 remains comparatively smooth. I suspect that's partly because there's no output transformer to add a layer of grit.

OG OC?

The OC818 was conceived, it seems to me, as the mic AKG would have made 50 years ago had the technology been available to manufacture the CK12 consistently and exploit its strengths fully. »

“I'd go as far as to say that this is the most effective shockmount I've ever tested.”



► Unusually, the shockmount does not use elastic cord; the inner cradle is suspended using triangular pieces of rubber-like material





SM-8



NOMINEE

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» As a consequence, it performs much better than the C414EB from a technical point of view, and does not have the same design or construction faults. It also does not reproduce some of the sonic characteristics that were, arguably, accidents of that design.

Something of the signature sound of AKG's CK12-based mics is undoubtedly down to the rather small Haufe T14/1 output transformer that was used in nearly all of them. This seems to add its own contribution to the sound, and I suspect it's partly responsible for the high-frequency sparkle that distinguishes

those mics. That's not present in the OC818, or at least, not in the same way, and the same goes for the OC-S10. If you want that slightly pushed, insistent treble, you'll have to manufacture it by passing the OC-S10 through a coloured preamp. Because, by itself, this is just a wonderfully natural-sounding mic. It doesn't have the ruler-flat response you'll get in a good pencil mic, but more importantly, it never feels as though the mic is coming between the source and the listener. Inasmuch as this is a flattering mic, it carries out its flattery in a completely invisible fashion.

ALTERNATIVES

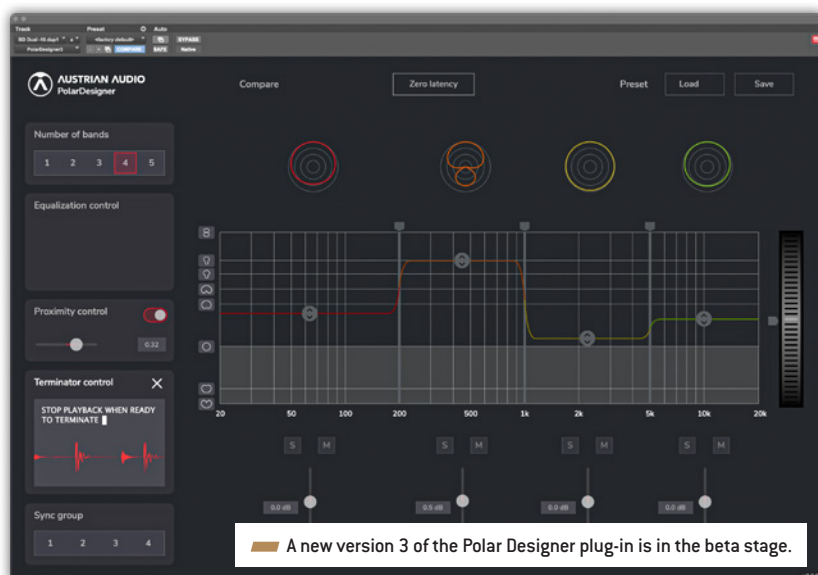
To my mind, the most obvious rival to the OC-S10 is **Nordic Audio Labs' NU-100K**, which is based on a new capsule design that seems to have a lot in common sonically with the AKG CK12.

The OC-S10 is a very fine microphone, and had Austrian Audio launched this rather than the OC818 back in 2019, I've no doubt that it too would have become an instant classic. However, given that the earlier mic has established itself so successfully, the question arises as to why you'd pay more than twice as much for a new model that, on paper, is very similar.

I didn't have an OC818 available for comparison, but I didn't hear anything from the OC-S10 to suggest that there's likely to be a huge sonic difference between them — and nor would I want there to be. Why would Austrian Audio choose to change a winning formula? However, the OC-S10 is a real improvement in other ways. The second full-size XLR turns the option of recording the two sides separately from something that's nice to have in theory into something you'll actually use in practice. The shockmount is an absolutely outstanding piece of engineering that offers better isolation from stand-borne noise than anything else I've come across. The pop shield is likewise one of the very best designs you can buy. And the whole package just has an air of understated class that will fit perfectly into today's boutique destination studios.

Would I buy an OC-S10 when I could get an OC818 Dual Set and have money left over for more mics? That's a tough question, and ultimately the answer to that would depend on how I anticipated using them. If I needed a pair of dogsbody mics to handle everything from drum overheads to brass overdubs, the 818s would be right at the top of my list. But if I wanted one statement mic to place in front of the most demanding vocalists, I'd have every confidence in the OC-S10 to do me proud. **///**

Plug & Play



By selecting the central (cardioid) polar pattern and connecting a second XLR cable, it's possible to record the outputs from the front and rear sides of the OC-S10's capsule separately. Assuming you're careful to match the gain across both preamp channels, the resulting two-channel recording can then be processed using one of three native plug-ins, all of which are available for free on Austrian Audio's website. These were developed for the OC818, but as far as I'm aware, should work equally well with the newer mic.

Polar Designer allows you to recombine the two channels to synthesize your chosen polar pattern after the fact. It goes much further than simply offering a single, continuously variable pattern, though. You can vary the polar pattern independently across up to five separate frequency bands, or ask the plug-in to automatically optimise the pattern so as to maximise capture of the wanted sound or minimise spill pick-up. It's also possible to reduce or enhance proximity effect.

In some circumstances, the two-channel output can also be valuable as a source of stereo information. In its raw form, the stereo track represents a coincident array of cardioid mics at a 180-degree mutual angle. This is not an oft-used setup, though there are times when it might get you out of a hole. In this situation, the second of Austrian Audio's plug-ins, Stereo Creator, allows you to vary the polar pattern of both sides simultaneously. It really comes into its own, however, if you have two coincident dual-output mics mounted at a 90-degree angle. By recording all four outputs to the channels of a quad track, you can then decode all sorts of different stereo arrays. Taking this further, AmbiCreator derives a B-format signal from the same mic setup, allowing you to use it as the basis for a two-dimensional Ambisonic array.

All of the plug-ins are elegantly presented and easy to set up, but it should be noted that until now, there have been no Apple Silicon native versions. However, Austrian Audio confirmed that these are in the works, and an open beta of Polar Designer 3 with Apple Silicon support and other enhancements is already available.

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— Universal Audio Volt 876

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The Hardware

At the heart of this studio will be the brand-new **Volt 876** interface from **Universal Audio**, providing a generous 24 inputs and 28 outputs via USB, including eight analogue inputs boasting 1176-style compression and a Vintage mode offering the sound of UA's classic 610 preamp. There's a comprehensive monitor control section, digital I/O via ADAT (and word clock for sync), plus full-size MIDI ports, and two headphone outs.

UA will also be providing two of their acclaimed guitar pedals. The **UAFX Anti** and **UAFX Knuckles** are based on classic '90s guitar amps, offering iconic rock and metal tones in a stompbox format. Also from UA is a complete drum recording setup, comprising an **SD-5** kick mic, an **SD-7** snare mic, two **SD-3**

and the realism players expect from top-tier V-Drums.

Danish microphone legends **DPA** will be offering two of their new **4099 Core+** instrument mics. Based on their established 4000-series capsule technology, the new Core+ version promise best-in-class noise performance, alongside unbeatable sonic transparency.

CAD Audio will also be providing two microphones: the **E100SX** large-diaphragm hypercardioid mic from their acclaimed **Equitek** range, and a **GXL800** small-diaphragm cardioid

model, which between them will handle almost any audio source.

Digital modelling experts **Slate Digital** are giving away their **ML-1A**: a quality mic in its own right, but when paired with Slate Digital's **Virtual Microphone System** software it offers accurate emulations of dozens of classic modern and vintage mics.

This year also marks a quarter of a century for microphone giants **sE Electronics**, who will be providing a limited-edition 25th anniversary edition of their acclaimed

Reflexion Filter Pro.

Rounding off the microphone collection is the **Warm Audio WA-44** — a ribbon microphone based on a classic American design. They'll be pairing that with their acclaimed **WA-MPX** valve microphone

To enter, please visit:
<https://sosm.ag/40th-anniversary-competition>

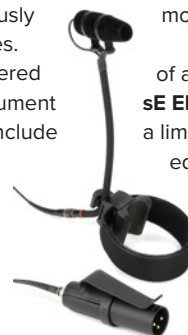
triple mic packs (six in total) for toms and hi-hats, and a pair of **SP-1** pencil mics for overheads. A further **SD-1** dynamic mic is ideal for capturing bass or guitar, while the large-diaphragm capacitor **SC-1** will take any vocal in its stride. All these mics are compatible with UA's pioneering **Hemisphere** modelling platform, offering a huge array of meticulously modelled microphones.

We've also partnered with electronic instrument legends **Roland**, to include their brand-new

V-Drum kit, the **TD516**. Whether you're tracking detailed performances in the studio or demanding more nuance from your live setup, the TD516 provides ultra-dynamic control, deep connectivity,



— Roland TD516



— DPA 4099 Core+ & Universal Mount

preamp, which pays tribute to the legendary Ampex 351.

Console legends **Harrison**, meanwhile, will be weighing in with the **32Classic Mix Strip**, a complete channel strip comprising a premium transformer-coupled mic pre, a four-band parametric EQ, high- and low-pass filters, and flexible routing options.

Monitoring will be well catered for, with both **ADAM Audio** and **KRK** contributing to this massive prize bundle. The **ADAM D3V** boasts the refined ribbon-tweeter sound we associate with ADAM but in a super-compact and portable format. For when serious low end is required, **KRK's Rokit 8 G5** offers LF extension down to 32Hz (-10dB) courtesy of KRK's trademark yellow Kevlar cone, and a high end that reaches up to 40kHz.

The winner will also receive not one but two pairs of headphones. **AIAIAI's TMA-2 Studio Wireless** is a closed-back moving-coil design with the unique ability to operate wirelessly, at low latency and with near-lossless audio quality. **Fostex**, meanwhile, are including their **T50RPMK4**, a planar magnetic set of headphones that boast exceptional transient performance, high sensitivity and ultra-low distortion.

Also celebrating an amazing 40 years in business this year are **Focusrite**, whose best-selling **Scarlett 2i2 4th Gen USB** interface will also come as part of this prize bundle. The 2i2 4th Gen offers a compact, portable yet high-quality means of capturing audio to a computer.

British console legends **Allen & Heath**, meanwhile, are bundling in their **ZEDi10** compact console, which combines premium analogue mixing of up to four mono and two stereo sources, plus musical 'British' EQ, monitor and effects aux busses, and a built-in 4-in/4-out USB audio interface.

We're also giving away three premium 500-series modules. **SSL** will be giving away their **E-Series EQ Module** and **E-Series Dynamics Module**, essentially packing the sound of an E-Series console



Warm Audio WA-44

channel strip into two 500-series slots. Meanwhile, **Cranborne Audio** are giving away a **Brick Lane** compressor — a unique design that uses PWM compression to offer the sonic flavour of almost any type of compressor ever conceived.

IK Multimedia are donating their **Tonex Pedal**, an advanced amp and effects modeller that not comes with a plethora of virtual guitar rigs, and even allows you to capture your own.

Finally on the hardware front are two software controllers. The **Novation Launch Control XL 3** offers hands-on control over every major DAW, as well as third-party soft synths, effects plug-ins and MIDI hardware.

The **Black BT SusEx Pedal**, meanwhile, is an innovative pedal-based foot-operated unit that combines wah-wah-like control with instantaneous switching, courtesy of an auto-sensing system that can tell where you foot is placed on the pedal and modify its behaviour accordingly.

The Software

Every modern studio needs a DAW, and with this prize bundle you get a choice of two: **Bitwig** are offering up their flagship **Bitwig Studio**, a multi-platform suite of production tools with a modern, intuitive UI that makes working with audio a breeze. Its powerful modulation options make it a sound designer's



ADAM Audio D3V

dream, while naturally it can also perform all the usual audio and MIDI production tricks.

SoundBridge will be providing a lifetime licence for their brand-new self-contained production ecosystem. **SoundBridge: DAW** is a musician-oriented platform with a streamlined UI that's aimed at fostering creativity and avoiding overwhelm, whether you're sketching ideas or mastering finished tracks. Also included will Soundbridge's **RitMix** virtual drum machine, based around an MPC-style pad grid and 16-step sequencer, plus their **reChord** arpeggiator and melody generator.

From **Native Instruments** comes their flagship instrument collection, **Complete**

15 Ultimate. This gargantuan suite brings together



Native Instruments Complete Ultimate 15

some of the world's most popular virtual instruments and effects, so that you'll never be wanting for a sound ever again — whether you're crafting a dancefloor filler or an orchestral movie score.

Sampling legends **EastWest** will be providing a full year's access to their flagship **ComposerCloud+** subscription, offering access to some of the world's finest sampled instruments, including the complete Hollywood, Fantasy and Symphonic Orchestra series, their extensive rock, pop and world instrument collections, vocal instruments, synth and keyboard collections, and much more.

From Swedish sample maestros **Toontrack** comes the benchmark-setting **Superior Drummer 3**, a comprehensive virtual drum instrument that offers meticulously sampled drum kits and other percussion elements.

Effects gurus **Soundtoys** will be providing their comprehensive **Soundtoys 5** collection, comprising their complete catalogue of plug-ins — from the Little AlterBoy pitch- and formant-shifter to their Decapitator saturation processor and EchoBoy delay, plus many more.

Finally, we at **SOS** will be delighted to offer the winner a year's free access to our complete music production ecosystem, **SOS For Artists**. Comprising full access to premium software and plug-ins, an AI-powered mastering service, a huge searchable sample database, and even the tools you need to publish and release your music, SOS For Artists will help you on every step of your musical journey. And as one last cherry on top of this enormous cake, the **SOS** team will offer the winner a free one-hour remote consultation on how to get their new gear up and running as quickly as possible.

To be in with a chance of winning this enormous prize bundle, simply follow the URL shown, and answer all the questions there, by Monday 26th January 2026. Good luck! **///**

Harrison 32Classic Mix Strip





JOHN WALDEN

iZotope's Ozone is one of the most long-standing and well-known software mastering suites, and we last reviewed it at version 11, back in *SOS* January 2024 (<https://sosm.ag/izotope-ozone-11>). Now, iZotope have delivered version 12. As you might expect from a mature piece of software like this, Ozone's existing feature set is both broad and deep. It's a powerful piece of software that offers more than enough functionality for experienced engineers to perform almost any kind of mastering processing. All that core functionality remains intact in Ozone 12, but we've explored it in previous reviews, so I'll keep this review focused on the highlights of this new release.

There are several tier options when purchasing Ozone, and I had access to the 'full fat' Advanced edition for this review. The other options in the Ozone family, Elements and Standard, provide subsets of the feature set at more modest prices. Elements is aimed at less experienced users who might want the software's Master Assistant process to do the heavy lifting for them, while the key difference between the Standard and Advanced versions is the number of processing modules included (14 and 20, respectively), and that Advanced also makes each of its modules available to the user as an individual DAW plug-in. (The iZotope website provides

iZotope Ozone 12 Advanced

Mastering Plug-in Suite

The latest version of Ozone promises improved AI assistance and adds some interesting mix-repair options to its powerful mastering toolkit.

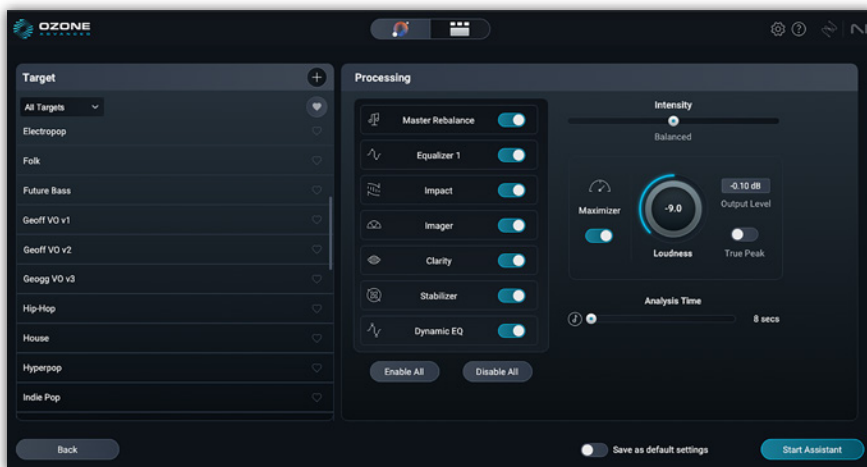
a feature comparison chart if you want more detailed information about the differences between the three tiers.)

A Little Assistance

Ozone's depth and power can make it somewhat intimidating for new or less experienced users, and it was with this in mind that iZotope first introduced the AI-based Master Assistant feature. Essentially, this auditions a section of the incoming audio and then suggests a suitable mastering processing chain to get you started, while leaving you in control. In Advanced and Standard, you can then dip into the individual modules, but in all versions — and lying at the heart of the Elements version — you can use a straightforward 'more/less'

approach, courtesy of a compact set of macro-style controls. No, this perhaps won't rescue a poor mix to the extent an experienced mastering engineer might be able to, but feed it with material in decent health and it's very effective, producing fast, fuss-free results that, importantly, make it less likely that an inexperienced user will take their mix backwards rather than forwards!

Version 12 brings us the option to customise this assisted mastering process. Sitting alongside what's now a much more expansive selection of genre-based targets, the new Custom feature lets you choose your own selection of Ozone's underlying modules to be included in the processing chain. You can adjust the length of the auditioning time



Custom signal flows and additional genre targets bring more options to the Master Assistant concept.

(up to 60 seconds, using the Analysis Time slider) that Ozone uses to learn the properties of your mix. And you can give the Maximizer module a nudge in your preferred direction in terms of the intensity of the processing and any 'loudness target' you might have in mind. Should you prefer a specific starting point, you can even save these 'custom flow' settings as a default — for example, this could be useful if you want to ensure consistency as you work through a series of tracks.

Some minor graphical tweaks aside, once the analysis is done, the options for then fine-tuning the result thrown up by Ozone 12's initial suggestions remain similar to those found in v11, and while this is not foolproof, I do find the compact control set to be well thought out. Indeed, it's what makes the Elements edition such a sensible choice for those who are new to

DIY mastering: it consistently delivers pretty solid results without the user having to actually learn how to master.

Know Your Limits

The Maximizer module is intended to increase loudness and add a sense of impact. I'll leave a discussion of the relative merits of squeezing every last drop of loudness out of your mix to another day, but how hard this processor is made to work is entirely within the user's control. If the user pushes for a louder end result, then management of Maximizer's limiting stage is key to how artefact-free the results remain. iZotope have added a new Intelligent Release Control mode (IRC 5) to this module's existing options and we're told

that it's their most advanced algorithm to date, able to deliver louder masters without trade-offs in terms of additional artefacts such as distortion or pumping.

As before, Maximizer's real-time graphical display provides useful visual feedback on the limiting process, and it's interesting to see the impact on this as you switch between the different IRC modes. Also interesting is when you engage the module's Delta switch as you do so — this isolates just the parts of the signal that the limiting is removing from the original. I've no insight into exactly how IRC 5 does its work (though the real-time graph hints that the limiting is faster) but it does seem, as advertised, to let you push the levels just a touch further while retaining the signal's coherence.

However, don't just assume IRC 5 is the best option in all cases. Like so many things in mastering, the metering is helpful, but you must let your ears guide your judgement, and I spent an interesting hour with a selection of mixes in contrasting genres — there were plenty of cases where I preferred alternative IRC modes.

Better Bass

Ozone Advanced already had a dedicated low-end management module in the form of Low End Focus, so I was intrigued at the new addition called Bass Control. This module has been added in both Advanced and Standard versions of Ozone. If you look at the documentation for the two modules, »

iZotope Ozone 11 Advanced \$499

PROS

- New modules expand Advanced's technical capabilities.
- Elements is an ideal starting point for DIY mastering.
- New Bass Control module is particularly impressive.

CONS

- A significant learning curve if you're to fully exploit the Advanced and Standard versions.

SUMMARY

Ozone pulls off quite a trick. From a technical perspective, Ozone 12 Advanced is surely the most powerful and sophisticated software-based mastering environment currently available, while in its Elements edition, it provides one of the most accessible.



The new Bass Control module provides impressive control over the balance and dynamics of the low-frequency range while mastering.

» their intended functionality appears to have considerable overlap, in that they both let you adjust the overall balance of the bass frequencies within your mix, while also giving options to modify the transient and sustain portions of the bass content. But when switching between the two, I did generally prefer what the new Bass Control module had to offer.

I found the new control set more intuitive for shaping both the overall bass level and the dynamics of the low end. Balance and Punch are definitely 'go to' controls, and the meters on both can guide your moves. However, alongside the oscilloscope display, the Sustain Power and Peak Control also let you adjust the low-end dynamics in some useful ways. Contemporary mixes are often not shy when it comes to the bass content but, applied to older mixes (for example, when remastering a project from last century!), Bass Control proved a neat way to improve the low-end impact, with apparently little or no compromise on the integrity of the audio. Low End Focus still does a good thing — but Bass Control does it better!

Re-bake That Cake

Mastering engineers have always been tasked with changing certain properties of the mixes they have been supplied with, but today they can be asked to do more, for example through stem mixing — a halfway house between mixing and mastering that allows mastering engineers more control without having to send things back to the mixer.

The addition of the Stem EQ module to the Advanced edition suggests iZotope want to provide mastering engineers with options to tackle more challenging stereo mastering projects in a similar way as they can in stem mastering. Stem EQ operates in 'real time' (ie. during playback, though there's a modest latency, which was 85ms on my host system). It separates your stereo mix into four stems: vocals, bass, drums and 'other'. And while no real-time separation can quite match the fidelity provided by offline processors (such as found in iZotope's own RX software), I felt that Stem EQ did a very respectable job during my tests.



■ Stem EQ lets you reach inside the mix to adjust the EQ of up to four stems in real time.

Like the existing Master Rebalance module, you can then adjust the relative gain of the stems, but as the 'EQ' part of the name implies, you can also apply equalisation separately to each stem. You have low- and high-pass, low- and high-shelf and bell (parametric) filters available, with (I believe — the documentation doesn't specify) up to eight EQ nodes available per stem.

OK, you could do a similar thing previously using the Stem Focus controls, but Stem EQ makes it much easier. It provides an impressive degree of control and, with the option to solo a stem also included, it's easy to focus in and hear what your EQ moves are doing. So, if the vocal needs to be louder and a little brighter, or the snare needs a touch more low-mid energy, you can now reach inside the mix and attempt the edit. Essentially, Stem EQ lets you un-bake the cake, adjust the amount of specific ingredients, and then re-bake it again.

Unlimited Options

The other new module added to Ozone Advanced is Unlimiter, and this also falls clearly within the 'fix the mix' category. As the name suggests, the module attempts to undo the results of overly aggressive limiting on the mix bus. As you might expect, it's based on some cutting-edge AI/machine-learning technology, and the module essentially aims to create the missing peaks of affected transients. As with Stem EQ, it's quite remarkable to see this happen in real time and, having set a threshold for the processing to operate from (or let the module 'learn' it for itself), you just dial in the Amount control and watch the peaks reappear within the graphical display.

In use, the Delta button is again very revealing, and the audible results of Unlimiter's processing can be pretty subtle in the grand scheme of an intense (ie. busy and loud) mix. But mastering is often about the combined effect of fine details, and if the ability to coax back a couple of dB to the drum transients goes some way to restoring a sense of dynamics, then Unlimiter will have done a useful job.

From a technical perspective, both Stem EQ and Unlimiter are doing remarkable things that are only made possible by the recent rapid advances in software-based audio analysis and machine-learning techniques. Both might arguably fall into the audio rescue and restoration category, and you might think they belong in RX more than Ozone. But I can see clearly how both could prove useful to an engineer who's asked to master a track where there is no recourse to return to the original mix. (I can also imagine the rolling of eyes and deep sighing that might be involved when said mastering engineer realises they are going to have to turn to Stem EQ or Unlimiter, because it's easier for them to 'fix' a mix rather than have a difficult conversation with an artist or mix engineer who's just delivered a sub-standard mix!).

Worth It?

What I've discussed in depth above are the main highlights of this new release, but there are a few other 'new for v12' features worth noting. These include additional genre targets in Stabilizer, a number of UI refinements, and improved AI/neural training for the stem separation processing that underlies Stem Focus and modules such as Master Rebalance and Stem EQ.

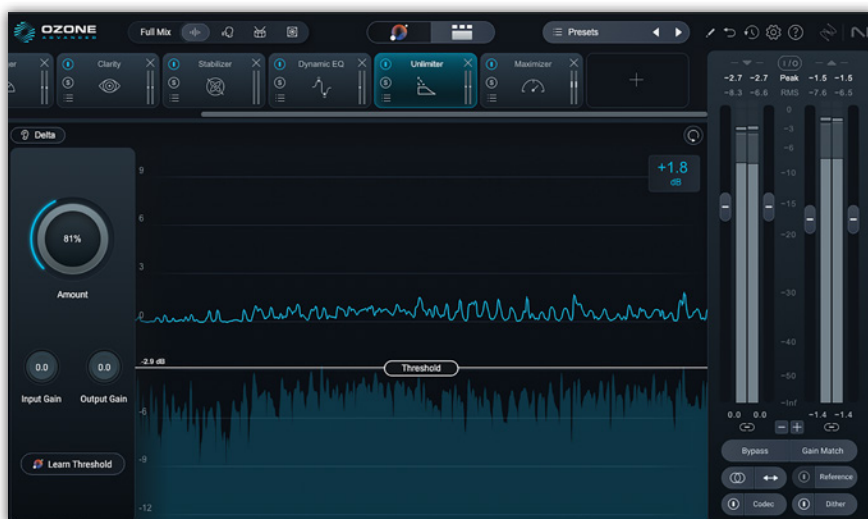
So, should you buy it? It's certainly capable of great results, and there are

some powerful new options here that move Ozone along in some very useful ways. Like most sophisticated and mature software tools, it's a case of evolution rather than revolution, but that's not a criticism — indeed, the existing user base should be thankful for that. It does mean, though, that

update decisions will depend very much upon whether you're likely to use any of the headline additions.

For potential new users, there's a lot more to explore and a lot to like. Ozone's three editions seem to me to be very clearly defined. Advanced is sophisticated,

powerful and deep, and intended for professional mastering engineers who are comfortable with an entirely software-based signal chain. It offers what is frankly an amazing array of tools. With its more compact selection of modules (and lack of access to those modules as individual plug-ins), Standard gives you much of that same experience but at a more affordable price. Finally, Elements is an ideal choice for those who need to undertake DIY mastering, but don't have the experience, skills or time to dive into the deep end with Standard or Advanced. Indeed, as a toe in the shallow end of the mastering pool, I think Elements is an excellent option. Personally, while it's perhaps not for the faint of heart, I love the power and flexibility offered by Ozone 12 Advance, which is undeniably as impressive as it is comprehensive. **///**



Overzealous mix bus limiting can now, within reason, be undone using the new Unlimiter module.

\$ Ozone 12 Advanced \$499, Standard \$219, Elements \$55. Upgrades available. Ozone 12 Advanced is also included within the Music Production Suite and Mix & Master bundles.

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PHIL WARD

I first wrote about Amphion monitors in these pages over 10 years ago, when I reviewed the One15 and One18 models (along with an Amphion power amplifier). That speaker has now been revised, with a new tweeter and a re-engineered crossover, to become the One18X.

The One18X is a relatively small monitor, so nearfield installation is not likely to provide any dimensional challenges. Its front panel carries a nominally 165mm-diameter bass/mid driver and a 25mm dome tweeter, the latter mounted at the mouth of a generous and matte-white curved waveguide. The waveguide is comparable in size with that of the bass/mid driver. Both the tweeter and bass/mid driver feature aluminium diaphragms, and they are integrated via a passive crossover at an unusually low 1600Hz. The One18X's advertised bandwidth is 45Hz to 55kHz ± 3 dB, and its sensitivity 85dB at 1m for 2.83V (1W at 8 Ω). It's specified to offer the amplifier a nominally 8 Ω load, but as we'll see, passive speaker impedance is a subject generously laced with complexities.

Around the back we find an auxiliary bass radiator (ABR), also equipped with an aluminium diaphragm, and some input terminals that can accept 4mm plugs, bare wires or spade connectors. The cabinet is finished in a dark grey textured paint and is constructed of internally braced MDF. It feels both extremely rigid and non-resonant. While I'm on the subject of cabinets, I've always really admired the classically 'Scandinavian' Amphion industrial design aesthetic. The combination of visual simplicity with classy finishes and beautifully judged proportions presses all the right buttons for me.

Passive Voice

Bar the One25A, all Amphion monitors are passive. I'm sure the vast majority of readers appreciate the difference between active and passive monitors, but just in case, here's a one-sentence explanation. Rather than incorporating multi-driver amplification and line-level (or DSP-based) crossover filtering within their cabinet, as do active monitors, passive monitors contain a crossover filter network that operates at power level, and are driven by external amplification. I'm not going to get into the pros and cons



Amphion One18X

Passive Monitors

With their precision engineering and distinctive white waveguides, the One series put Amphion on the studio map. And now they've got even better.

of the active and passive principles now, except to say that we in the pro-audio sector have come to expect monitors to be active, and perhaps to see passive monitors as maybe equivalent to a slightly weird relative: that uncle who lives out in the middle of nowhere with seven cats and is only ever mentioned in hushed

tones. But of course, just as Uncle Colin might well be living his best life and having a fabulous time, passive monitors can work extremely effectively — as evidenced by countless very satisfied Amphion users. Furthermore, wearing my speaker engineer hat, I have a soft spot for passive monitors; to my mind there's

Amphion One18X

\$4100

PROS

- Remarkable level of midrange and tweeter detail.
- Fabulous, low-distortion high frequencies.
- Pin-sharp stereo imaging focus.
- Very well judged bass.

CONS

- None.

SUMMARY

The Amphion One18 always was a very capable nearfield monitor. It's significantly better now and hugely impressive.

more creative artistry inherent in their design. Active speakers, from a design perspective anyway, can seem a touch 'paint by numbers' I feel.

Another Amphion feature is the generous tweeter waveguide. In electro-acoustic terms, the waveguide is perhaps more significant than the passive format, because it defines some of the electro-acoustic design parameters at a more fundamental level. A decade ago, when I first wrote about Amphion monitors, big tweeter waveguides were relatively unusual. These days, though, there are many more such devices to be seen adorning the front panels of monitors, which suggests the arguments in their favour have gained some traction. So what are the arguments?

Guiding Light

Any conventional two-way speaker, in which the full audio band is covered by a bass/mid driver and a tweeter, potentially incorporates a challenge: engineering a bass/mid driver that extends high enough in frequency to integrate with a tweeter that extends low enough is inherently difficult. If the bass/mid driver has a diaphragm large enough to reproduce low frequencies at a reasonable volume level, it will become significantly directional at the top end of its band; and at said top end of the band, a large diaphragm will have stopped moving as a whole and be working in a somewhat chaotic, resonant mode. The tweeter issue is that if the diaphragm is small and light enough to reach up to 20kHz with acceptably wide dispersion, it might struggle to reach low enough in frequency to integrate with the already bandwidth-challenged bass/mid driver whilst retaining

acceptable power handling and distortion levels. Furthermore, even when a bass/mid driver and tweeter are successfully integrated, there is likely to be a significant dispersion mismatch between them. The dispersion of the tweeter at the bottom of its band will be inherently wider than the dispersion of the bass/mid driver at the top of its band. The result of that will be potentially audible non-linearity in the off-axis response.

Adding a waveguide to a tweeter offers a potential fix. Firstly, the waveguide modifies the inherent radiation impedance mismatch between a tweeter diaphragm and the air, and the result of that is a very significant increase in tweeter radiation efficiency. A waveguide can add around 10dB to tweeter sensitivity, and that allows its operational band to be extended very usefully downwards because its overall workload is significantly reduced. Secondly, tweeter dispersion, rather than being defined by the diaphragm diameter, will be defined at lower frequencies by the waveguide diameter, and if that is comparable to the diameter of the bass/mid driver, they will

have similar dispersion. So the dispersion mismatch between bass/mid driver and tweeter goes away.

A further benefit is that the increased sensitivity of the tweeter enables the crossover frequency to be lowered. This both relieves the bass/mid driver from duty at the top end of its range where the going gets tough, and, again, helps with the dispersion mismatch, because the bass/mid driver is low-pass filtered earlier (where it is less directional) and where the wavelength is longer (so path-length differences between the drivers and the listener result in less phase difference).

Loading...

The final signature Amphion characteristic is their use of auxiliary bass radiators for bass loading (the term 'loading' comes from the idea that you 'load' the bass driver diaphragm with the mass and compliance of the air in the cabinet). The ABR technique is closely related to 'simple' port tube reflex loading (ABR is in fact a form of reflex loading), but rather than employ a tuned port to extend the bass driver output, a driver diaphragm and suspension (with no motor system) is used.

To my way of thinking, there's two primary advantages to the ABR technique over the simple reflex port. Firstly, the ABR tuning frequency can be adjusted simply by adding or removing mass to or from the diaphragm. This is in contrast to having to adjust the length of the port, which, in compact monitors, might not fit easily in the cabinet once it is of large enough diameter to provide adequate linear airflow. Secondly, ABRs are immune to the potentially audible undesired resonance effects that simple port tubes often suffer. That perhaps makes it seem as if anybody would be crazy to choose the simple reflex port option, but of course simple ports do have advantages; they're inexpensive, simple (the clue is in the name) and consume much less cabinet panel real estate than ABRs, and with thoughtful design simple ports can be made to work perfectly well.

Gen X

And so to the modifications that have turned the One18 into the One 18X. The changes primarily concern the tweeter and crossover. Firstly, the tweeter is now the same as the device employed in the active high-end One25A that I reviewed, very positively, back in *SOS* »



Instead of a port, the One18X loads its bass driver with an ABR, or auxiliary bass radiator, mounted on the rear of the cabinet.

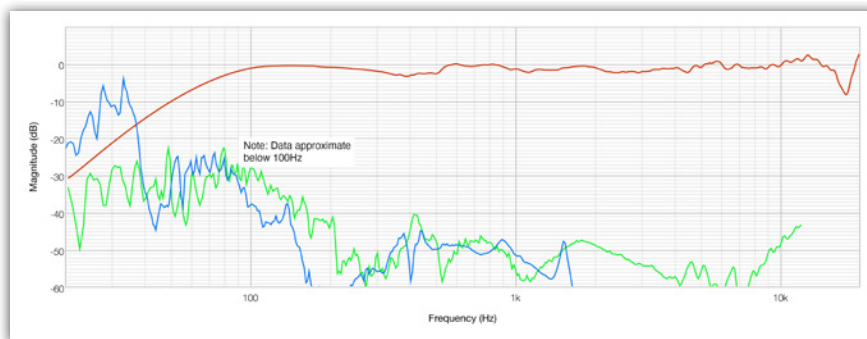
» May 2024. The revised tweeter offers, say Amphion, significantly extended high-frequency bandwidth, increased linear dome-excursion capability and much lower distortion than the 'old' tweeter. Interestingly, Amphion's founder, Anssi Hyvönen, says that part of the motivation behind developing the X tweeter is that the increasing prevalence of headphone listening and monitoring is raising listeners' expectations of extended HF bandwidth and lower distortion — both of which are 'easy' to achieve with headphone drivers. Along with the new tweeter the One18 also features a revised waveguide profile.

Any change to a driver in a passive speaker is going to require some crossover revisions. In the One18X, those changes are firstly aimed at accommodating any baseline response shape or sensitivity characteristics that the new tweeter brings. And, while improved objective performance is the goal, it was important to Amphion that the subjective character not be too different from the existing One18. Otherwise, qualities that made the One18 so popular would be lost.

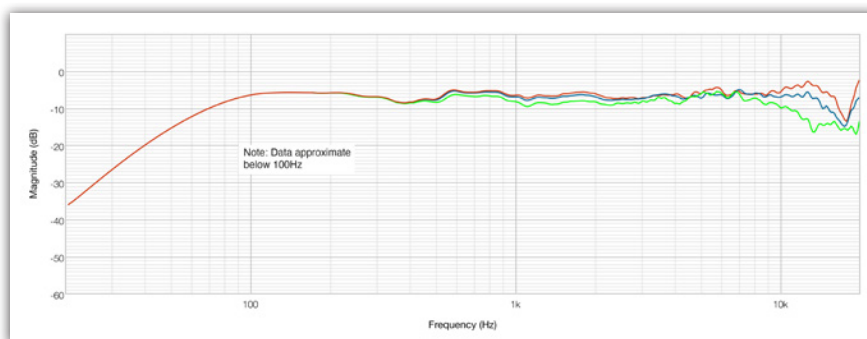
Measure Boogie

I took the One18X to my usual speaker measuring space and recorded some acoustic data that, hopefully, will illustrate some of the concepts I've been describing. Diagram 1 shows the One18X axial frequency response overlaid with second and third harmonic distortion curves (SPL was 90dB at 1m). While not the flattest axial frequency response I've ever measured, the One18X response is perfectly reasonable and generally falls within ± 2 dB limits between 100Hz and 10kHz. And, as I've written many times, there's so much more to any speaker than any single frequency-response curve. The One18X distortion performance is generally very good, but its high-frequency third-harmonic performance is notably excellent. It falls off the bottom of my graph and is typically better than 60dB below the fundamental, which makes it less than 0.1%. That's not unheard of in speakers, but it's not heard very often. Amphion's claim that the new tweeter offers lower distortion looks to be realised.

Returning to the axial frequency response for a moment, the dip and recovery apparent above 15kHz is sometimes typical of waveguide tweeters.



■ Diagram 1: The One18X's one-axis frequency response [90dB SPL at 1m] is shown in red. The green and blue traces show second- and third-harmonic products, respectively.



■ Diagram 2: Comparing the axial response [red] with measurements taken 20 and 40 degrees horizontally off-axis [blue and green traces, respectively].

It's caused by the inherent narrowing dispersion of the naked dome as frequency increases (the dome radiation 'decouples' from the waveguide). Some speaker engineers combat the effect by placing a high-frequency dispersion-modifying 'phase plug' in front of the dome, while some feel that this kind of technique risks introducing audible artefacts. I'm agnostic on the matter, and

"It's the kind of monitor that doesn't leave you second guessing..."

it's a perfectly reasonable decision on Amphion's part to leave the phenomenon unmodified — 15kHz and above is, after all, questionably significant for the majority of listeners.

Diagrams 2 and 3 provide a perfect illustration of the midrange dispersion control that comes with the combination of a waveguide tweeter and a low crossover frequency. Diagram 2 shows the One18X axial frequency response overlaid with curves for 20 and 40 degrees horizontal dispersion. The off-axis curves are extremely impressive in their uniformity and lack of any major discontinuities. This is especially the case through the midrange, where

a conventional, non-waveguide system would more typically show a significant off-axis midrange dip followed by a recovery on the other side of the crossover region.

Diagram 3 shows the axial frequency response again, overlaid with curves for +15 and -15 degrees vertically off-axis. While the -15 curve shows a small degree of phase cancellation in the crossover

region, the +15 curve remains nicely linear through the crossover. The practical result of this curve is that the One18X is best heard either directly on axis or slightly above. If the monitors were installed on a relatively high monitor shelf, I'd consider turning them upside-down.

Finally, before I get to describing what I thought of the One18X subjectively, Diagram 4 shows its input impedance and, as with the vast majority of speaker impedance curves, it illustrates that amplifiers don't have an easy time. I've overlaid a third curve to Diagram 4 in addition to the usual magnitude and phase curves. This Equivalent Peak Dissipation Resistance (EPDR for short) curve displays the effective resistive load the amplifier "sees" from the combination of magnitude and phase. The One18X

»

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» EPDR curve drops to under 5Ω for a significant portion of the audio band, and while that's relatively low, it's not nearly as amp-punishing as some EPDR curves I've seen. In the grand scheme of things, the One18X constitutes not too challenging an amplifier load. Earlier I quoted the One18X impedance spec of 8Ω. Is that fair? Well, traditionally (I believe it dates back to a 1970s German DIN specification), to qualify as an 8Ω speaker the impedance shouldn't drop below 6.4Ω anywhere in the audio band. I think the One18X just about squeaks through — its minimum is just over 6Ω at 150Hz.

Amped Up

Firing up the One18X and loaned Amphion Amp700 power amplifier, and listening to a variety of reference tracks and in-progress mixes, my first impressions were of a slightly bright high-frequency balance combined with a very slightly warm midrange, almost as if the One18X has a faint smile shape to its tonal balance. But I was also struck by the way the One18X reveals immense quantities of detail in the midrange and lower-HF bands: unintended noises-off, edits and reverb tails that I'd never been quite so aware of before in material that I thought I knew well became more

explicitly audible. Stereo imaging, too, was of a hugely high standard, with focus and depth presentation that extends beyond just the central phantom image.

Monitors have a noise floor, where all the various distortions and cabinet panel, driver or port resonances skulk around, and if the noise floor is low, then more programme detail becomes audible and stereo imaging becomes more solid. In those terms, the One18X undoubtedly has an extremely low noise floor, and that helps to make it a superb monitoring tool. To my way of thinking (and listening), a monitor's noise floor is more important, within reason, than its tonal balance — not least because these days, with such easy access to monitoring EQ (via room optimisation or monitor control software), monitors can have pretty much any broad-brush balance we like. And having said that, as the One18X's raw tonal balance was marginally bright for my taste, I spent the rest of my listening with a 1dB shelving cut from 4kHz upwards implemented in my Audient ORIA interface control app.

With or without my tonal balance preference, in addition to offering immense detail in its presentation, the One18X tweeter handles sibilance really well, without a hint of harshness or sounding pushy. There's no sense

ALTERNATIVES

Considering an approximate package price including an Amphion Amp700 power amp, active monitors worth considering in the same kind of ballpark could include products such as the **ATC SCM20A**, **MesanoVIC CDM65**, **PMC6**, **PSI Audio A21M** and **Genelec 8341**.

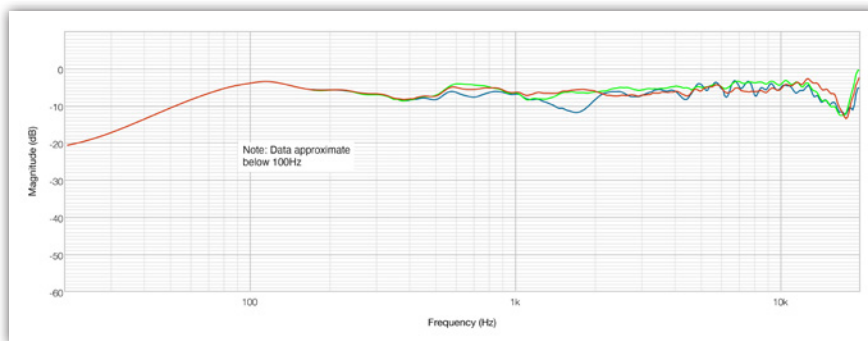
of, "Hey! Listen to me! I do detail!", just an accurate and fully realised version of what's present in the programme material. I think there are two phenomena at play here. Firstly, the well-managed mid and high-frequency dispersion of the One18X system means there's no wild swings of tonal signature between direct and reflected sound reaching the ears; and secondly, the incredibly low distortion generated by Amphion's new tweeter, especially in the third harmonic, means it's adding almost nothing objectionable of its own.

I've always admired the ABR-based balance Amphion have developed between the temporal accuracy of closed-box bass loading and the bandwidth and power-handling advantages of reflex loading. The impedance curve of the One18X actually reveals that its low-frequency alignment is relatively well damped, and in electro-acoustic terms, I see it living somewhere between closed-box and reflex; successfully expressing some of the characteristics of both techniques. And that's just how One18X bass sounds. Its bandwidth is nicely extended, so it can play some very low bass, but that bandwidth extension isn't achieved at the expense of time or pitch accuracy. It's the kind of monitor that doesn't leave you second guessing what's going on between that kick drum and bass guitar. Again, that makes the One18X a really effective mix tool.

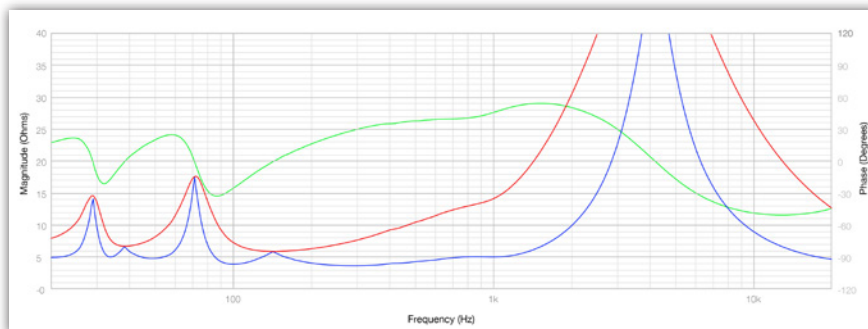
To wrap up, I'm really pretty certain that the One18X is an improvement on the One18, and my feeling is that it's not a small improvement either. Both my measurements and listening suggest that Amphion's new tweeter is a genuine advance, and its effect lifts the One18X performance to a new level. The One18X is one of those review monitors that I was sad to see packed up and waiting for collection. ■■■

\$ \$4100 per pair.

W www.amphion.fi

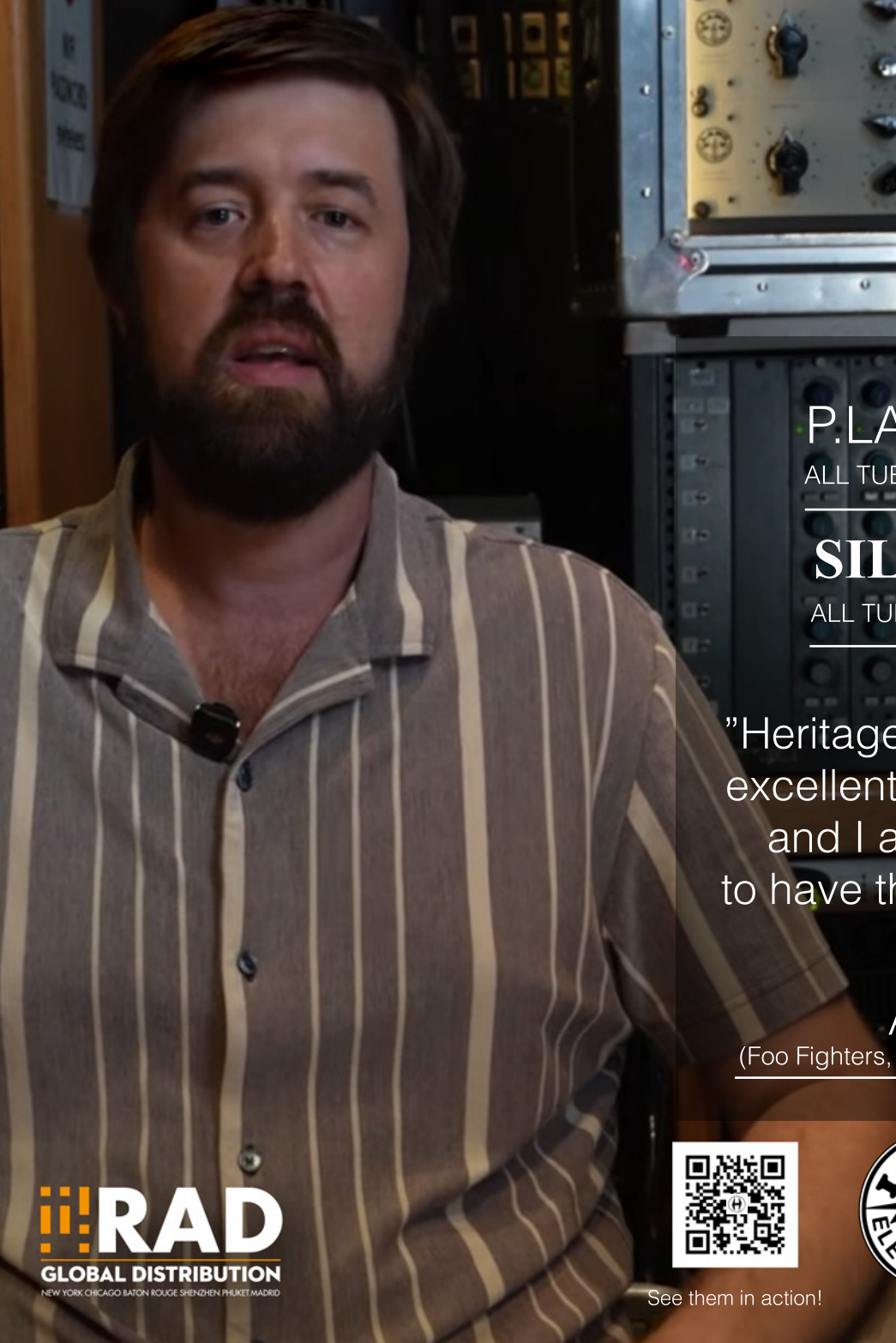


■ Diagram 3: Axial response (red) versus +15 and -15 degrees vertically off-axis (green and blue traces, respectively).



■ Diagram 4: The One18X's input impedance. Red = magnitude; green = phase; blue = EPDR.

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SONAL D'SILVA

American composers the Newton Brothers have spent the past decade and a half terrifying audiences in creative ways with their scores for projects like *The Grudge*, *Five Nights At Freddy's*, *The Devil's Hour*, *Ouija: Origin Of Evil*, and more. With the release of *Tenebra*, in collaboration with Spitfire Audio, they share the tools of their trade to help composers conjure up all manner of nightmares, one meticulously-sampled menacing note at a time.

With a download size of 11.35GB, *Tenebra* runs in Spitfire Audio's Solar plug-in, powered by their eDNA (Electronic DNA) technology, and contains more than 100 sounds, each of which was "born from experimentation", according to the literature. Orchestral instruments, field recordings, human voices, percussion, modular synths and unconventional instruments are processed and ready, providing the building blocks for a suspenseful composition.

Overview

A great place to start, before exploring the 130 presets on offer, are the 10 factory presets that give you an overview of the core sounds that this library was built on. This is good for new composers (to learn what the foundation sounds like), and experienced composers (who often like to modify sounds from scratch). Here, you will find the full range of sounds that *Tenebra* offers: from staples like glassy, piercing tones (Bells & Glass) evolving, textured drones (Drone), impactful percussion (Perc) and creepy human sounds (Humanity) to more unique sounds from the composers' personal synth collection (Synths), the dark side of an orchestra (Orch-Nocturna), and the Newton Brothers' signature sounds "mangled by the Spitfire Audio team" (Warps).

This is just the tip of the iceberg because after the factory presets come the other presets, with creepy but vague names like *Angry Bees Dancing*, *Docking Off Jupiter*, *Freddie's Basement*, and *Here & There*, for example. I must give SA a shoutout from a usability perspective: if you mouse over the information icon to the right of each name, you get a clear sonic description of each patch. *Angry Bees Dancing* is a "buzzy pulsing synth", *Docking Off Jupiter* is a "gritty atmospheric bass", *Freddie's Basement* is a "dark airy texture", and so on. This is incredibly helpful, especially during the early days of using the library

Spitfire Audio Tenebra Sample Library

Spitfire have teamed up with composers the Newton Brothers to create something that's truly terrifying.

when you're not yet familiar with it and just want to get to work instead of wasting time interpreting what *Here & There* could possibly mean for your composition. (It's an "eerie wailing texture", FYI.) A tip for new composers: when exploring the library, play the short audio preview next to each name and then read the corresponding text description. Sounds can be very hard to articulate sometimes and this is good practice for becoming more precise with your own descriptions.

Sounds & Textures

The Synth presets are a treat, featuring sounds created using an ARP 2600, Moog Matriarch, DFAM, Polygogo, Therevox, Oberheim OB-X, Folktek Resonant Garden, Super Gemini, Solar 42 and the Newton Brothers' expansive Eurorack modular system. You have complex textures, gritty leads, arpeggiated basses and eerie pads, and that's before you even begin manipulating the sounds.

Another standout is the Strings, with slippery glissandos, unhinged risers, chaotic swells and terrifying ghostly bends. One of the highlights of *Tenebra* is that while it's built around experimentation, it's not so esoteric that it abandons what we've come to know as the core sounds of a horror and suspense score, a solid string component being one of them.

The Bass category has two interesting presets: *Graveyard Haunting Bass* featuring *Purgatory's Cello* as sound A and *Red Flute Flutters Kit* as sound B. As you go

higher in pitch, the timbre changes to something bordering on sound design. It sounds chaotic, like everything is going crazy, so you can progress in tension within one patch itself. Terrifying *For Huge Chords Pad* is also interesting because the higher notes turn the dark, gritty bass into an alarm-like sound. This is a useful characteristic seen across the library — textures and sounds within one patch vary as you travel around the keyboard.

The library also boasts a selection of some unusual instruments including a bowed medieval citole, a "perfectly imperfect bowed piano", and the

»

Spitfire Audio Tenebra \$199

PROS

- Large selection of character-filled samples.
- Detailed and precise sonic manipulation.
- Easy to link parameters to MIDI CC.
- Light on CPU resources.

CONS

- None.

SUMMARY

Tenebra by the Newton Brothers is a rich library filled with evocative presets and extensive effects for customisation of the sounds. From horror, thriller and suspense staples like chaotic swells, eerie drones and dread-inducing bass to unusual instruments, unconventional performances and complex textures, this is a library that you will get more and more out of as you grow as a composer.

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Welcome to the Modal Era.

» Waterphone, an iconic horror instrument developed by musician and composer Richard Waters, whose sound many of us have wrongly assumed to be that of a particularly creepy violin. It's a delight to have access to some of its varied textures and performances.

Effects

Half the fun of Tenebra is in the samples it contains, and the other half is in the engine that runs it. Solar is described as being the most up-to-date iteration of the eDNA engine, originally created for Spitfire's Earth library, and now expanded. It is a powerful interface that allows almost too many options for sonic manipulation. I say too many only because it can be intimidating at first. If there's something you want to tweak, it's likely to allow you to tweak it.

Each preset consists of two sounds — one each in sound bays A and B — with individual controls per sound for the following parameters: wobble (volume, pitch, filter); ADSR envelope; high- and low-pass filter; tune, pan, offset, trim; and glide, bend and clone.

The two sounds in conjunction can be manipulated via the Oscillate Mixer, which crossfades between bay A and bay B. Turn it on and it moves on its own, but you can also manually control its movement via the mod wheel. The crossfader multiplier increases or decreases the speed of the Oscillator Mixer and you can also select

the shape of the oscillator, speed, direction strength, and start point of the oscillation.

This is a useful feature in a preset featuring two distinct sounds, but it's even more fun to turn it on with two identical sounds loaded. I tried it with two instances of a sound called Moms Chimes and it created a sort of vacuum in the centre of the image field that was more felt than heard but was distinct enough to make a difference. On headphones, it generated the sensation you get when your eyes are closed and someone holds a finger close to the middle of your forehead. A horror library awakening your third eye? That's not spooky at all!

While the Oscillate Mixer facilitates crossfading between two sounds, what happens if you want to control the individual volume of each sound? That's where the Trim knob comes in. You can balance the levels between the two sounds to your liking, and can even use it to fade in one sound after the other one has been triggered. One more layer of creativity unlocked.

Bays A and B have identical parameters on the main page, but when you go deeper into the FX section, you notice the difference between the two. Each of the bays has a pre-assigned effects rack that differs a little from the other. Bay A has Shape, Phonic, Peak, Amp, Distort, Flanger and Chorus. Bay B has Form, Sonant, Notch, Digital, Bias, Convo and

Phaser. Both have aux sends where you can control Delay, Reverb, Convolution, Chorus, Flanger and Phaser. At the end of the signal chain are the Master FX including EQ, Grain, Tape Saturation, and a Limiter for final polish. Another shoutout for the excellent Tenebra manual, which features a clear and easy-to-understand eDNA FX signal-flow diagram so you can understand exactly what the signal path is and adjust effects accordingly.

The final section to explore is the Gate Sequencer, which is simple to understand. Two rows of step sequencers control both sound bays independently, and you can mute or unmute each step to add rhythmic movement to the sounds. Several parameters are adjustable including the length of the sequence, and the speed, depth and shape of the gate. A cool feature is that you can swap the sequenced patterns between the two bays easily using the Flip icon, to test which pattern works better with which sound.

Conclusion

There's no place more fitting for 'vibes' — the defining characteristic of present day pop culture — to shine than the world of horror music composition. Something either feels terrifying or it doesn't and arguing about the technical merits of a composition will do you no good if it has left your audience cold. Tenebra is so good with vibes that it almost feels like cheating — hold down a chord and the sounds seem to do the work for you.

Online forums are littered with people who scoff at sample libraries that have famous names attached to them. For example, the oft-echoed sentiment that "buying a Hans Zimmer library is not going to make you compose like him!" Indeed. The point being missed here is that the best sample libraries that involve famous composers offer us a selection of their musical vocabulary, and, by extension, access to instruments and sounds that you would not otherwise have access to, in one place. Think about it: as a non-famous, non-wealthy composer, would you have access to an orchestra, a rare medieval instrument, a Waterphone, vintage synths, and a selection of haunting glass tones all in one place? Unlikely. If the price works for you, this is a wonderful toolkit of well-curated sounds to inspire your next creepy creation. **///**



■ The FX knob at the top can be configured to control multiple parameters. Alt+right-click on any parameter to assign it to the knob.

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Arturia Tape J-37

Tape Emulation Plug-in

In recent years, French company Arturia have steadily been growing their collection of effects plug-ins, and I use some of them on my mixes quite regularly. In particular, I like their spring reverb and EHX Memory Man emulation, but when I heard legendary analogue tape aficionado Eric Valentine describe the release I'm writing about here as the best software tape plug-in he'd come across, my interest was truly piqued — I have a lot of respect for Eric's work, and he still maintains and uses his vintage tape machines.

I won't attempt an in-depth history of the Studer J37 here, but it's safe to say this one-inch, four-track, valve-amplified open-reel tape machine has left its mark in the history books of recording technology, not least through its connection with the Beatles' *St Pepper* sessions. Its modular design was innovative at the time, and it employed more than 50 vacuum tubes. The specific machine that Arturia chose to model spent most of its working life in Studio Barclay, which Arturia describe as the "Abbey Road of French music". Using a "neural-based modelling framework and conducting painstaking multi-stage measurements", Arturia claim to have captured the "exact interactions, harmonics and 'tape states' that define the Studer J37's distinctive sound".

Arturia's Tape J-37 plug-in sports a good-looking GUI with a nicely laid-out selection of controls, and there's enough control under the hood to permit a useful amount of tweaking too. The Color section is perhaps the most significant, offering a sonic choice ranging from a 'pristine' tape emulation, with 'new' SM911 tape loaded onto the machine, to SM468 tape at various stages of its life cycle. All these options can be driven into saturation using the Drive control, but the grittiest and dirtiest options will take you more into outright disruption territory — in a good way, if that's the effect you're after. Despite some jumps in level that sometimes made critical judgment a little tricky, I always found it enjoyable switching between these options. Unlike with many tape emulations I've used, you can hear clear, audible differences, so it's easy to make quick decisions and



move on. That this plug-in doesn't feature a wet/dry control may frustrate some, but really, I found that the Drive control was all I needed and didn't find that a problem in use — and you could always use it as a send effect, of course.

The EQ section plays a big part in how effectively this plug-in performs, and the ability to place it either before or after the tape component offers some subtly distinct tonal options. Hitting EQ before the tape, for example, could sometimes seem more forgiving. I particularly like the Emphasis options, which allow you to steer the obvious tape saturation effects toward or away from, for example, the low-end.

The Delay section adds yet another layer of creative tinkering, and is intended to mimic the well-known tape-echo effects that were created with such machines before digital effects came along. I found myself using this section more and more during the review period, in fact; it has just the right amount of control options to get things just as you want while you quickly move through a mix.

Finally, I was pleased to note that the Noise and Wobble options seem to be turned off as standard! But they can be easily brought back in to taste if you're looking for a more authentic model of an older or poorly maintained tape machine.

I found myself agreeing wholeheartedly with Eric Valentine's assessment of this plug-in — put simply, I've used a lot of tape emulation plug-ins, and Tape J-37 is the most effective I've heard to date. It's a plug-in I can happily put on a guitar or drum bus and immediately find more agreeable, in a way that's really hard to articulate. Then there are the more creative options for adding more obvious tape character or delay effects. All in all, this plug-in is definitely worth considering, and there's a demo available to help you decide. *Neil Rogers*

\$ €99 (about \$115).

W www.arturia.com

Brainworx bx_pulsar

Delay Plug-in

Brainworx's latest offering is bx_pulsar, and just as bx_aura does for reverb, bx_pulsar can handle conventional delay treatments while also providing some very interesting creative options. The topmost toolbar (which includes a useful Test Tone button for auditioning your settings) aside, the GUI is divided into upper and lower panels, and at the top-centre, you can select from five different main delay modes: Air, Magnetic, Circuit, Aether and Reverse. Air provides clean digital options, Magnetic offers different flavours of tape-based delay, Circuit emulates bucket-brigade delay (BBD) circuits, Aether includes creative 'radiance' (blends in a pitch-shifted 'octave up' layer) and 'sparkle' (applies grain-based processing to the delays), while Reverse does what is says on the tin. The controls arranged around this central mode selector change with the mode but, amongst other things, for Air, Magnetic and Circuit, they include modulation options, while both Aether and Reverse include an interesting Diffusion option that blends reverb with the delay signal to smooth the repeats out.

Top-right, you get both low-/high-cut and low-/high-damping filters that can be placed before or after the delay line. More conventional delay controls are found at the bottom centre of the GUI, with the usual delay time, sync, mix, feedback and stereo width options included. Brainworx have followed the very welcome current trend of including a ducker (bottom right) — this tucks the delay signal out of the way when the source signal is present, letting the repeats come up more in between, which can help to maintain the clarity of the source. Interestingly, this ducking can also be driven via a side-chain.

With the exception of the pristine Air mode, all the other delay modes could be considered 'creative', and that's particularly true of the Aether and Reverse. However, three additional features ensure bx_pulsar does more than just tick the 'creative' box: the Freeze button, the Pattern section (top left) and the Groove section (bottom left).

Freeze does what you'd expect, with endless repeats created from the current contents of the delay line (and progressively more modulation applied if those settings are non-zero). Groove



lets you apply both swing and/or a timing offset to the delay line. The latter allows you to push/pull the repeats in relation to the beat in some interesting ways, while the former lets you apply swing to the pattern of repeats. It can be a cool effect if you want to subtly enhance the swung nature of a groove.

The Pattern section is probably the icing on the creative delay cake, though, and two modes are offered. Dual mode provides dual delay lines with four different setups, including ping-pong and criss-cross. The stereo nature of the delay can be enhanced using the L/R Offset setting, which can shift the timing offset between the channels up to a maximum of 50ms. The Ratio control is also particularly interesting. This allows you to generate different numbers of repeats with the left and right delay line in the same timespan. For example, a Ratio of 2:3 means that the left delay line repeats twice over the same time that the right delay line repeats three times. A number of different Ratio settings are provided, and there's real potential here for some cool, trippy, rhythmic effects. Sequence mode is more straightforward in concept — you activate any/all of the eight steps, and then adjust their respective panning — but it has just as much creative potential.

“I can’t recall the last time I had quite so much fun with a dedicated delay plug-in.”

Brainworx have supplied a good crop of presets to demonstrate many of the possibilities across vocals, drums and other instrument types. Conventional delay treatments are well covered, of course, but it's the more creative options that really make bx_pulsar stand out, and I can't recall the last time I had quite so much fun with a dedicated delay plug-in. Sonically, the five distinct delay algorithms cover a lot of ground and the utility elements — the filter and ducker in particular — are very useful. The GUI is slick, the main controls easy to navigate, and those creative options are very cool. It's also very fairly priced; even at the full price, bx_pulsar is accessible, but existing Brainworx users also qualify for a discount. Yes, I already own more delay plug-ins than I need... but nonetheless, I've a feeling bx_pulsar is going to see a lot of use here! If you have room in your own plug-in folder for one more delay option, then bx_pulsar would be well worth checking out. *John Walden*

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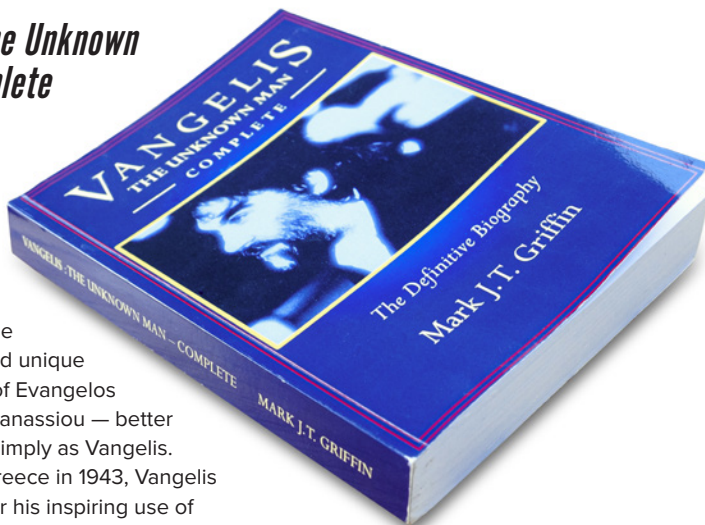
Vangelis: The Unknown Man – Complete

Book

It's hard to conceive of anyone not being aware, at least on some level, of the extraordinary and unique musical talents of Evangelos Odysseus Papathanassiou — better known to most simply as Vangelis. Born in Agria, Greece in 1943, Vangelis is best known for his inspiring use of synthesizers in creating hundreds of solo and collaborative works released as singles, albums and soundtracks throughout a career that spanned 1962 to 2022. We all know the *Chariots Of Fire* and *Blade Runner* soundtracks, of course, and many will be familiar with his electronic albums such as *Heaven and Hell*, *Albedo 0.39*, and *Juno to Jupiter* (his very last album). But how many are aware that he also wrote significant classical compositions for ballet, theatre and concert orchestra?

Mark HT Griffin's unofficial biography (*Vangelis: The Unknown Man — Complete. The Definitive Biography*, ISBN 978-0-9533017-6-8, published by Gryffyn Productions Ltd) stems from a long-held interest in and passion for Vangelis' work. The original 90-page incarnation of this book appeared in 1994 before being expanded to 176 pages in a 1997 second edition. But this new self-published version spans 400 pages, and is clearly the culmination of a massive effort to detail the complete life and career of Vangelis.

The content is drawn from vast swathes of material, derived mostly from countless published articles and interviews in music magazines and newspapers, printed throughout the extent of Vangelis' public popularity, as well as from numerous album sleeve notes, TV and radio documentaries, and a variety of websites which materialised long after the earliest versions of this book. Although there's nothing official from Vangelis himself (or his representatives) as far as I can see, some of his collaborators and friends have contributed to what is a very thorough and comprehensive collection of information, stories and anecdotes,



describing the composer's life and works in a fascinating and entertaining way with considerable detail.

While many of the synthesizers and other equipment used by Vangelis and his associates in various productions are mentioned casually in passing — and there is an appendix listing equipment known to have been used during different periods of his career — there's little technical detail concerning this equipment, its use, or the production techniques Vangelis developed and employed. Consequently, this book is more of historical interest than detailing the musicological or innovative technological aspects of his work.

I was sent a pre-publication book for review, and I have to say I found the typography pretty dense and hard on the eye due to the constant switching between Griffin's own commentary in a normal font, and italicised sections for quoted material. But happily that doesn't detract from the captivating story of one of my own musical heroes!

The book is divided into 18 key chapters, each covering a period in Vangelis' life defined mostly by a specific project. The book also includes many rare and private photographs, and several further appendices provide extensive listings of Vangelis' recorded albums, soundtracks, collaborations, and even his album sales statistics. There is also a full bibliography and references list. *Vangelis: The Unknown Man — Complete* is well worth reading by anyone interested in the works of Vangelis, not least for its gathering of so much information into one cohesive story. *Hugh Robjohns*

\$ £26.99 (about \$37).
W www.markjtgriffin.com

Warm Audio Tube Squealer

Overdrive Pedal

Overdrive pedals are a staple of live performance for guitar players, but they can also be beneficial in the studio — a good analogue pedal placed before a modelled amplifier can often sound and, importantly, 'feel' better to the player than using modelled pedals. Warm Audio's latest is the Tube Squealer, and with its name and green case that slopes down slightly at the front, there's little doubt as to what it's based on! The thing about Tube Screamers, though, is that they went through several design iterations over the years, and even though they all shared the mid-focused character of the original TS808 (which dates back to the late 1970s) each version sounded slightly different. The circuit has inspired many other pedal designs too. The Ibanez and Maxon originals most often employed a JRC4558D dual op-amp and a couple of clipping diodes, augmented by some tone-shaping and buffering circuitry.

The Tube Squealer has external switches for true or buffered bypass, and for activating a charge pump to bring the internal operating voltage up to 18V for more headroom. For those not using a PSU, there is a battery compartment accessible via a removable plate on the bottom of the pedal (do take care not to lose the two screws!). Where this pedal differs from most Tube Screamer clones is that it can be switched to emulate a TS808, a TS9 or a TS10, which it achieves by reconfiguring its internal circuitry to match the originals. It also has a wet/dry Mix control as well as Drive, Level and Tone controls. Furthermore, it has a two-position pickup type switch that voices the pedal for use with single-coil or humbucking pickups.

The three versions of the Tube Screamer were all based around the same circuit topography but with some component changes. The general character of the original's drive is warm, mid-heavy and quite compressed-sounding, and to my ear it has the smoothest sound of the three. Original TS808s are now very collectable and consequently fetch high prices, of course. Come 1982 and the TS9 was launched — I had one back in the day — and this was apparently the model favoured by Stevie Ray Vaughan. It had a slightly less 'polite' sound than the TS808, adding just a hint more bite. During the mid-'80s, the TS10

“The pickup type switch adds some top to the humbucker setting, and if you like your overdrive extra bright, you could use this setting with a single-coil pickup instrument.”

appeared on the scene, and was famously adopted by John Mayer, which certainly helped push up prices of secondhand models! Its sound was similar to the TS9, but again subtly different.

Fortunately you don't need to decide which one is best, because in the Tube Squealer you have the option to switch between all three types, with the added benefit of a Mix control — a simple touch but a welcome one, as it means you can blend in a little of the clean sound

to further preserve attack and clarity. The pickup type switch adds some top to the humbucker setting, and if you like your overdrive extra bright, you could use this setting with a single-coil pickup instrument. To my ear, the sound is close enough to its inspirations that even hardened 'diode sniffers' would struggle to find issue with what this little pedal serves up, and it is solidly built too with a cast metal case.

Paul White

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ART Pro Audio Pro Channel III Channel Strip

With switchable input impedance, dual compression voicings and a semi-parametric EQ on board, the Pro Channel III certainly offers tonal versatility. Could it be your next 'front end'?

MATT HOUGHTON

Founded in 1984, ART Pro Audio now seem to have been delivering competitively priced preamps, processors and other audio products for as long as I can remember, but it strikes me that in recent years a number of their rackmount products in particular have really grown into maturity. We looked at their Solo MPA & Solo VLA — the former a preamp, the latter a compressor — about a year ago (*SOS* November 2024: www.soundonsound.com/reviews/art-pro-audio-solo-mpa-solo-vla), and fellow reviewer Bob Thomas seemed pretty much blown away by the combination of sound and facilities for the price, noting as his only 'con' the lack of side-chain filtering on the compressor. ART have now combined elements of both devices with an EQ section to create a full recording channel called the Pro Channel III. It's not distributed in the UK/EU (yet?), but ART indicated to me that they're willing ship to international customers.

Overview

The Pro Channel III comes in a black, 2U 19-inch rackmount steel chassis, whose

front panel is divided visually into four sections, and whose profusion of knobs each turn detented pots. The construction quality seems decent to me, and the finish on the front panel is similarly pleasing. For example, on the review unit the detents all lined up nicely with the panel labels and other markings, which isn't always the case with more affordable outboard.

The leftmost section is the preamp, of course, and next come the EQ and compressor in that order but, usefully, their position in the signal path can be reversed. Finally, a 'master section' comprises a large VU meter with a recessed calibration pot, and this meter can be switched to show the input or output level. Below this, a 10-LED gain reduction bar meter for the compressor comprises one yellow and nine red LEDs. And to its right, a master output level control is labelled from minus infinity to +15dB. Actually, this doesn't *quite* give you full attenuation, but there's certainly a useful practical gain range. As well as setting the main output level, this knob serves as the make-up gain stage for the compressor. Doubling up in this way is an understandable cost:benefit decision, but since the master output knob remains

active when the compressor is bypassed it's not quite as easy as with some other units to quickly compare the compressed and uncompressed signals.

Around the back, you'll find a balanced input XLR socket, and the balanced XLR output is paralleled on a TRS jack. There's also a button to set the operating

ART Pro Audio Pro Channel III

\$500

PROS

- Clean-sounding preamp and EQ.
- Genuinely useful compressor options.
- Very versatile.
- Separate insert send/return.
- Good metering, with front-panel trim.

CONS

- Compressor lacks side-chain filter and dedicated make-up gain control.
- A dedicated balanced line input would be nice.

SUMMARY

A decent-sounding recording channel, the Pro Channel III is definitely a cut above the more budget options out there, and boasts some impressive features for the price, not least the choice of optical or FET-based compression.



level to +4dBu or -10dBV. The maximum input levels are +22dBu in the +4dBu setting, and +13dBu for -10dBV, while the maximum output level is +26dBu.

Unusually, for what's a relatively affordable device, there are also two TS jacks providing separate (unbalanced) insert send and return points. Obviously, the insert point means you can patch, say, a de-esser or gate/expander in between the preamp and compressor/EQ stages, but the return point can also be pressed into service as a line input. That could be useful, since although I found the preamp could cope with line-level signals, some might prefer not to risk exposing line-level devices to phantom power, or may prefer to have the line input available separately from the mic one on a patchbay. This way, you're bypassing the mic/instrument preamp when you patch into the insert return, so you'll have no input gain control and will need to manage the input levels with the previous device in the chain. But it's still a handy option. Finally, there's a standard IEC C13 power inlet with integral fuse compartment for the internal power supply, and this accepts 110-240 V and 50/60 Hz AC.

Stages

The FET microphone preamp has knobs for gain (0-40 dB) and a wide-ranging sweepable high-pass filter (10-250 Hz). There's also a +20dB gain boost switch (which doubles as a clip LED), which is rather like having a Cloudlifter-style booster preamp on board, and this takes the total input gain range to 0-60 dB.

That's not the full story, though, as the master output level control gives you a further 15dB of gain (so 75dB in total). What there isn't here is any pad, but thankfully that didn't seem necessary — even when hitting a snare drum hard with a sensitive mic placed close to the batter head, I never encountered anything the preamp couldn't cope with.

Further switches engage +48V phantom power and select the input impedance, the options being 600Ω (useful for some vintage mics, and potentially coaxing a different character from passive dynamic mics) and Max, which at 47kΩ should present a suitable load to any mic. There's also an 800kΩ front-panel instrument jack that takes precedence over the rear-panel mic inputs. And while we're on the subject of input impedance, the aforementioned insert return presents a 7.5kΩ load, while the unit's output impedance is given as a suitably low 47Ω. The final control in this section is the master on/off rocker switch — I approve of these being on the front panel of rackmountable devices, rather than tucked around the back!

While there's no polarity inverter in the preamp control section, you'll find one with the adjacent compressor controls. Here, you also have knobs for the familiar compression parameters, again all fully variable across their range: threshold (-30 to +20 dB), ratio (a very gentle 1.25:1 up to a near-limiting 20:1), attack (0.25 to 50 milliseconds) and release (0.15 to 3 seconds). Those ranges are all pretty generous, so allow you to configure the compressor for pretty much

any application. Beneath, alongside the polarity inverter, are three more switches: Voice, which switches between FET and valve-amplified optical compressor circuits; Comp, which switches the chosen compressor in/out of the signal path; and a button to toggle the order of the compressor and EQ.

Which brings us neatly to the EQ section, whose low and high shelves and two swept mid bands combine with the preamp's high-pass filter to give you five bands in total with which you can shape the sound. Each of the four main bands has a ±12dB gain knob. For the 12kHz high shelf, in fact, that's all you get, but it's ample for brightening or taming the top end, and of course its gentle curve reaches further down into high mids than its nominal turnover frequency. The low shelf, on the other hand, has a switch to toggle the turnover frequency between 40 and 120 Hz, while the two central bands have generously variable frequency knobs, ranging from 20Hz to 2kHz for the low-mid band, and 200Hz to 20kHz for the high-mid. So there's plenty of overlap, and that last one can operate well beyond the mids, should you want more control over the high end than is offered by the HF shelf alone.

Generally speaking, this emphasis on versatility is very welcome, as it means the Pro Channel III really can be useful on pretty much any source. The cost of those sweeping ranges is that the controls can be slightly sensitive in operation: small turns can sometimes make a big difference. Still, there seemed to be a pleasing physical resistance, and it really wasn't difficult in practice to dial in my desired settings. There's no control over the EQ bandwidths, but they seem to be a proportional-Q type — the greater the boost/cut, the narrower the curve becomes — and sound pretty natural to my ears. The final control for this section is an in/bypass switch for the EQ stage.

In Use

That concludes our tour of the facilities, so what's this thing actually like in use — and how does it sound? I tried it with a range of mics, including moving-coil dynamics, both passive and active ribbons, and a couple of capacitor mics, to capture a number of sources that I felt should reveal any strengths and weaknesses of the Pro Channel III: vocals, acoustic and electric guitar, electric bass guitar, a tambourine and a snare drum. »



As well as the XLR mic input, there are balanced outputs on TRS jack and XLR, and there are separate TS jacks for the unbalanced insert send and return.

» The preamp sounded decent to my ears. Not something that adds an obvious character, and not something that I really enjoyed driving into distortion, but it came across as clean and detailed, and I never had difficulty finding the necessary gain. Importantly, that includes the minimum gain, which as I say allowed me to capture very hard snare-drum hits, close-miked with a sensitive capacitor mic, without problem. The switchable VU meter also made it really easy to dial in the right gain, as well as set the master output level. It perhaps felt a little strange looking to the ‘compressor’ section for the polarity inverter, but that’s not exactly a problem. The passive pickups of my bass guitar and a couple of regular electrics also seemed happy running into the instrument input; as with the mic preamp I never struggled for level or noticed an obvious character being imposed.

The preamp’s clean character is, of course, complemented by the various onboard tone-shaping options. The single-pole (6dB/oct) high-pass filter’s smooth slope is intended, say ART, “to remove rumble, pops, and wind noise”, and it did so successfully, but since it’s continuously variable up to 250Hz, I found it equally useful in compensating for proximity effect, and for working in tandem with the EQ section to shape the bottom end and low mids of various sources. So full marks yet again for versatility here. Some might wish the HPF could be switched out of the channel path entirely, but while its lowest 10Hz setting may cause a small phase-shift, it’s as good as off for all practical purposes.

I like to shape things with EQ and compression ‘on the way in’ — it may be fashionable to record clean and put processing off until later, but I’d much

rather get as close to the right sound as I can at the outset as, even if you err on the side of caution, it generally makes the job of mixing easier down the line. Outside of the boosts and cuts you dial in, I wouldn’t say there’s huge character on offer in this EQ stage; it’s pretty clean-sounding, in the way a console-style EQ should be. But any EQ can have a big tonal impact on a signal, of course, and this one is really quick and intuitive in use. The layout feels familiar, and with just six knobs and one switch to control four bands, you’re never overwhelmed. Certainly, for example, I was able to dial

“The insert point means you can patch, say, a de-esser or gate/expander in between the preamp and compressor/EQ stages, but the return point can also be pressed into service as a line input.”

up the breathiness in a vocal, to brighten ribbon mic recordings, cut some boxiness from a kick, and to push up the mids nicely for various miked guitar cabs. Nothing that wowed me, particularly, but it’s not to be sniffed at: a capable workhorse EQ. The detents and labelling mean it’s pretty easy to jot down and recall settings too.

It’s when you bring the compressor into play that you can really start to coax something more than workmanlike service out of the Pro Channel III. Its two flavours impose different characters and curves onto proceedings, with the valve-amplified optical circuit generally sounding smoother, more rounded and more forgiving than the FET one, which has the ability to act in a much grabbier, more assertive fashion if you wish. I could often simply switch from one to the other and quickly discern which I most preferred for a given source — it was easy to hear that I generally preferred the tube/opto variant on my bass guitar, for instance — but since they behave so

differently, you’ll likely need to tweak the threshold, attack and release, if not always the ratio. Either way, I enjoyed having both options on hand, as each proved preferable on different sources.

It was also great having the option to place the EQ pre or post the compressor. As well as allowing you to attend to any tonal problems before compression, such as the proximity effect, as I mentioned above, it can be great fun to, say, crank the mids on an electric guitar before it hits the compressor.

There are only a few tiny limitations worth highlighting. First, that lack of dedicated compressor make-up gain: to restore any level lost through compression, you must use the master output level knob, and as I said, this remains active

when the compressor’s bypassed so you can’t easily A/B compare the dry and compressed signals. There’s that point Bob Thomas mentioned about the lack of filtering in the compressor’s side-chain too; I’d love it if the HPF could optionally be re-routed for that. It would also be nice to have a preamp input for balanced line-level signals that could, separately, be brought out to a patchbay.

But given the price of this thing, that all feels like nitpicking! The bottom line is that the Pro Channel III is an incredibly versatile recording channel, and one whose combination of sound and features mean it punches well above its price. If you like to get hands-on with your processing but aren’t looking to spend thousands, this could well be the DAW ‘front end’ you’re looking for. **///**

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**Spaces are limited. Registration of interest does not guarantee an invitation to attend.

With an understanding of spaced and coincident stereo arrays, we can exploit the characteristics of both — using either established arrays, or by creating our own!

Part 6: Hybrid Stereo Mic Arrays

This is Stereo!



HUGH ROBJOHNS

In previous parts of this series, we've explored the concepts of coincident and spaced stereo mic arrays. I hope you'll recall that while coincident arrays generally deliver stable and precise stereo imaging, they're often described as lacking 'spaciousness'; in contrast, spaced arrays excel in the 'spaciousness' department but their stereo imaging is typically vague and unstable.

There's no technical reason, though, why these disparate approaches can't be combined to create a 'hybrid' array — one which uses directional polar patterns set with a mutual angle, but with the mics spaced apart. Indeed such stereo mic arrays are common and popular, and it's probably fair to say that most recording engineers use them most of the time! As with all the other stereo arrays we've considered, there are no fixed rules; the recording engineer can employ any capsule spacing, polar pattern and mutual angle they think appropriate for any given situation, bearing in mind that these parameters interact to determine the Stereo

Recording Angle (SRA) of the array, as well as the overall character of sound.

As with coincident mic array setups (crossed cardioids, Blumlein, Mid-Sides and so forth), there are many precisely defined and named hybrid mic arrays, most being devised by European broadcasters in the early 1960s and consequently named after those organisations.

Popular Hybrid Stereo Arrays

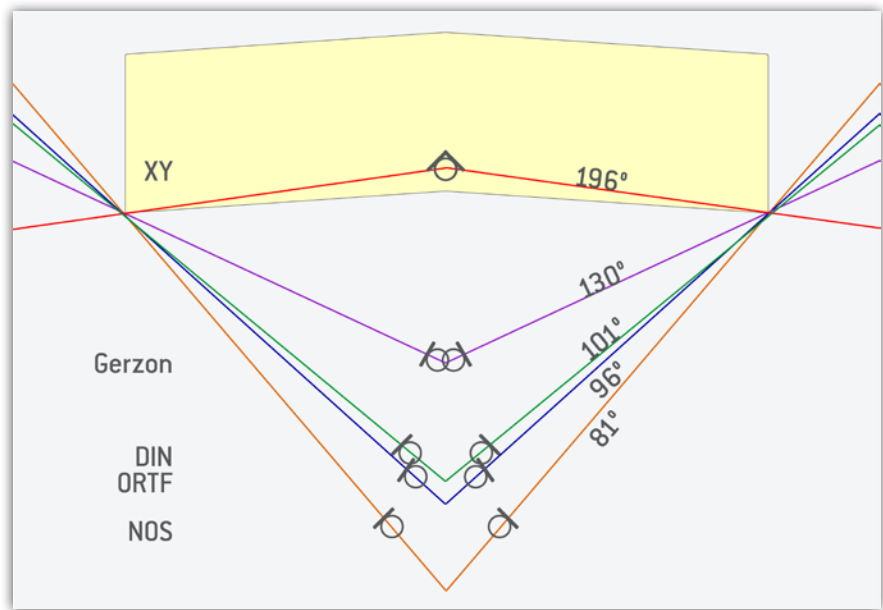
The best known and most popular is the ORTF mic technique, which was developed around 1964 by the Office de Radiodiffusion Télévision Française (the post-war French state broadcaster, which was broken up into seven new broadcasting companies in 1974, including TF1, TF2, FR3, and Radio France). For this array, two cardioid capsules are spaced apart by 170mm, and face outwards with a 110-degree mutual angle to give an SRA of 96 degrees. At around the same time, in the Netherlands, the national broadcasting foundation (Nederlandse Omroep Stichting) came up with a variation on the same theme. This NOS configuration also uses cardioid capsules but they're

arranged with a 90-degree mutual angle and a 300mm capsule spacing, a combination that gives a slightly narrower SRA of 81 degrees. And in Germany, the Deutsches Institut für Normung standards institute came up with the DIN array. This also employs cardioids with a mutual angle of 90 degrees, but its slightly narrower capsule spacing of 200mm gives a wider SRA of 102 degrees.

There are virtually limitless combinations of different capsule spacings, mutual angles and polar patterns, so not surprisingly there are other less well-known named formats, such as RAI (cardioids, 210mm spacing, 100-degree mutual angle) and EBS (cardioids, 250mm spacing, 90-degree angle). I've skipped over these simply because they're so similar to the arrays I've already described, but I will highlight one more option, since I plan on returning to it in the context of post-production signal processing in a later article. The Gerzon array employs cardioid capsules with a mutual angle of 120 degrees and a spacing of just 50mm, giving an SRA of 130 degrees. This array was named after and promoted in the 1980s by Michael Gerzon, a genius who's probably best known as the inventor of (first-order) Ambisonics and the SoundField microphone, and who credited the array to award-winning recording engineer and producer, Tony Faulkner. I wrote a detailed article about it five years ago, which you can read at www.soundonsound.com/techniques/gerzon-array.



Diagram 1: The tighter an array's SRA, the further back you must place it to have the source occupy the full stereo soundstage.



To help you appreciate the practical differences between all these named arrays, Diagram 1 (top right) indicates a notional orchestra on stage (the yellow block) with each hybrid stereo array positioned such that its SRA encompasses the full width of the orchestra. Since these arrays all use cardioid capsules, I've also included coincident X-Y cardioids, with their nominal 196-degree SRA.

As can be seen, the NOS array's relatively narrow 81-degree SRA means it must be placed furthest from the orchestra, with the X-Y coincident cardioids being the closest (typically right above the conductor). Consequently, each of these arrays will deliver a similar stereo width when auditioned on speakers, but very different perspectives (direct/reverb balance) due to their different distances from the stage; the more distant placements capture more ambience and reverberation than the closer ones, but also have different degrees of 'spaciousness' due to their differing capsule spacing. Again, NOS offers the greatest spaciousness and X-Y the least.

Practical Mic Placement

But faced with so many options, how do you choose? As with coincident and spaced arrays, the basic principle of mic placement is to decide first on the desired perspective — how far away from the source(s) the mics should be for the optimum balance of direct to reverberant sound. When rigging and rehearsal time allows, you can assess that by listening

to a single mic (of the type you intend to use for the stereo pair) facing directly forwards, with the monitoring switched to mono. Then adjust the placement (distance and height) to capture a sound with a nice overall balance of the ensemble on stage — but make it very slightly too dry, since it will sound more reverberant in stereo.

Having established the optimum position for the array, you can figure out the Stereo Recording Angle needed to encompass the entire ensemble. I'm not great at estimating angles by eye, so I use a digital protractor made by GemRed (I'm sure other brands make similar devices). I stand where the mic array is to be located, and open the protractor's arms until they align with the

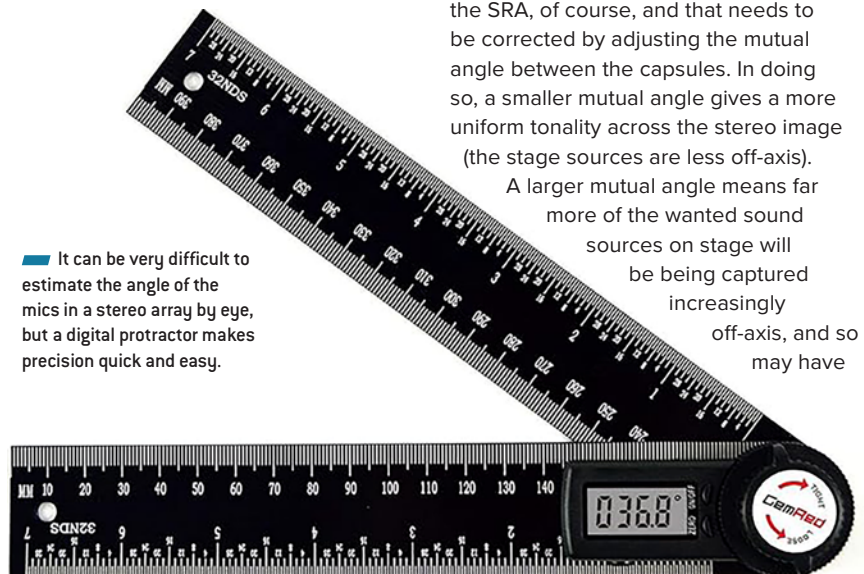
outer edges of the wanted soundstage while looking over the hinge. The display in the protractor gives me the exact angle required, and I can then refer to a stereo mic visualiser app to explore the various combinations of mutual angle and capsule spacing for my chosen mics' polar pattern that match that SRA value.

Choosing a smaller capsule spacing means more of the stereo information is conveyed as inter-channel level differences, which means sharper imaging detail but less spaciousness. Selecting a wider capsule separation generates more inter-channel time-of-arrival information, so a greater sense of spaciousness but less image stability and precision.

Changing the capsule spacing alters the SRA, of course, and that needs to be corrected by adjusting the mutual angle between the capsules. In doing so, a smaller mutual angle gives a more uniform tonality across the stereo image (the stage sources are less off-axis).

A larger mutual angle means far more of the wanted sound sources on stage will be being captured increasingly off-axis, and so may have »

It can be very difficult to estimate the angle of the mics in a stereo array by eye, but a digital protractor makes precision quick and easy.



» a less consistent tonality. This depends on the consistency of the mic's polar pattern with frequency, and I'd generally avoid large mutual angles when using large-diaphragm microphones for that reason, but when using good-quality small-diaphragm mics I find that it's rarely a noticeable issue in practice.

A third variable, especially when using multi-pattern mics, is to change the capsule polar patterns. For example, changing from cardioid to subcardioid has a similar effect to reducing the mutual angle — because a sound source must be further away from the centre to generate the same inter-channel level difference. Swapping cardioid for hypercardioid is akin to increasing the mutual angle, and

narrowing the SRA. This extra option means we can add a third rule:

Rule 1: Reducing the Mutual Angle increases the Stereo Recording Angle.

Rule 2: Reducing the Capsule Spacing increases the Stereo Recording Angle.

Rule 3: Reducing the Pattern Directivity increases the Stereo Recording Angle.

Again, it's important to remember that all of these parameters interact to some extent, so the ideal mic placement and array configuration is often arrived at iteratively. So my

advice is to always allow yourself plenty of time to work through that process, auditioning the results at each step to make informed decisions and choices. Yes, the perspective and stereo imaging can be fudged somewhat in post-production, but it's always far better to get it right at source, which is also the *only* way to arrive at the best possible sound quality.

With experience, it becomes easier to intuit an appropriate array close to the ideal from the outset. But building that experience takes a lot of time, experimentation and critical assessment in many different recording situations and environments! Recording is a craft and an art — it takes time to hone these skills.

Next Time

In the next part of this series, I'll discuss in more depth the practical considerations when choosing hybrid stereo arrays, examine the concept of 'angular distortion', and explore the potential of more sophisticated stereo arrays using more than two mics, including the famous Decca Tree. **///**

“Changing from cardioid to subcardioid has a similar effect to reducing the mutual angle — because a sound source must be further away from the centre to generate the same inter-channel level difference.”

Audio Examples

The media files associated with this series (<https://sosm.ag/this-is-stereo-media>) include an example of an ORTF hybrid mic array (track 8, 'ORTF'). This was captured using a pair of Rode TF5 microphones supported on a 3D-printed clip designed specifically for that purpose and mounted on the same stand that also accommodated the Mid-Sides array (track 7, 'Coincident Mid-Sides'). The latter makes a useful comparator for coincident versus hybrid arrays — but remember to adjust the Mid-Sides decoder to match the decoded stereo image width of the ORTF pair. Also mounted on the same stand was a pair of spaced omnidirectional mics (track 11, 'OCCO Omnis'), with a capsule spacing of 690mm for an SRA of 96 degrees — exactly the same as the ORTF array. Consequently, these spaced omnis allow a direct comparison between a pure spaced array and the hybrid ORTF setup.

Listening to the differences between these three alternative arrays, in terms of stereo imaging, spaciousness and perspective can be very illuminating. Unfortunately, changes in tonality across the sound stage are almost impossible to perceive from these examples because of the relatively strong reverberation. But in drier acoustics, and with a wider sound source (for example, a full orchestra) those differences would be more apparent between the different arrays. Sadly, I didn't have sufficient mics or recorder channels available to rig DIN, NOS and Gerzon hybrid arrays too, but hopefully this ORTF example will give a flavour of the benefits of hybrid arrays — and I would certainly encourage anyone interested in stereo recording to experiment with different configurations.

As mentioned, the ORTF, NOS and DIN arrays are very precisely specified. For example, if the capsules aren't exactly 170mm apart and facing 55



Thanks again to the Teme Valley South Churches Choir (TVSCC), who repeatedly performed 'If Ye Love Me' by Thomas Tallis to help us demonstrate the characteristics of various stereo mic arrays.

degrees left and right, then it's not an ORTF array. Rather, it's a hybrid array of your own design. But if you find that 170mm spacing with a mutual angle of 120 degrees works nicely in a particular situation, there's absolutely nothing wrong with using that combination. Just don't call it an ORTF array!



NOVA

ONE COMPLETE SOLUTION

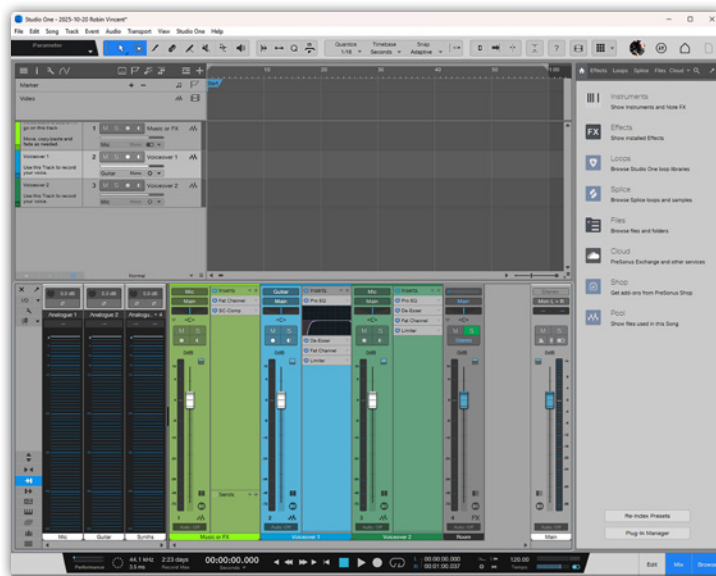
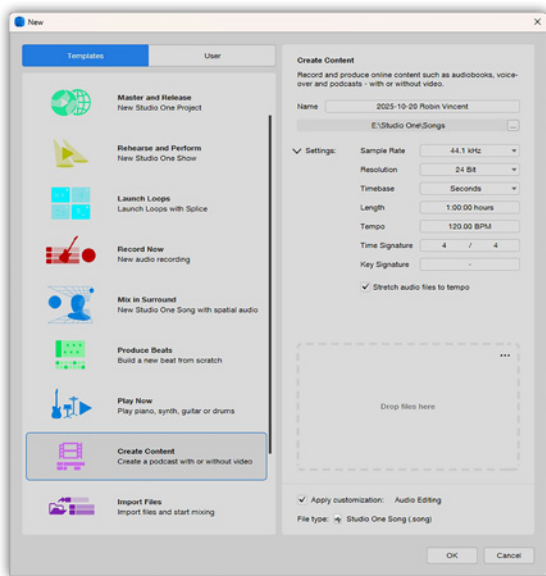
To take your monitoring to the next level



SOS EDITORS' PICK
GEAR OF THE YEAR 2025

www.trinnov.com

Studio One has everything you need to produce professional-sounding podcasts.



ROBIN VINCENT

Podcasts like to lean into the idea that they are recorded pretty much live, like radio, with minimal editing or post-production fixing. That's seldom the reality, though — and with a little bit of preparation, Studio One can reorient itself into a very efficient podcasting studio. In this workshop we'll look into how to make best use of everything Studio One has to offer in the production of your next podcast.

Templates

To kick us off, let's load up the podcasting template, which presents us with a ready-made recording environment. Click on New / Template and the Create Content option. This template loads up three tracks: one for music, and two for recording voices. The template also loads up the Marker track with a helpful 'Start' marker at the beginning, and opens the Video track (many podcasts are filmed at the same time for uploading to video-based sites such as YouTube).

A couple of notes on video. You can't record video into Studio One; you can drag video in afterwards, but you'll need a way to align it with the voice tracks you recorded into Studio One. The simplest way to do this is with a clap. With the camera and audio recording, clap your hands a couple of times. That sharp explosion of audio will be recorded both through your vocal mic and the mic on the camera or phone. Once imported into Studio One you'll be able to easily see and align the claps on the recordings. Simply drag your video into the Video track and click the little waveform

■ The Create Content template will load up a new session with everything you need to produce a two-person podcast with music.

symbol. This will extract the audio from the video to a separate, but connected, audio track. You should be able to see how the waveform display in the video's audio track matches up with your directly recorded audio. Finding the sharp peak of the clap will allow you to align your audio to the video more or less perfectly.

Recording Setup

The first thing to do is to set up the recording space. We're going to assume you're using two microphones to record a podcast with two people chatting together. Ideally you'd want to treat this as a vocal recording in a recording studio, but that's not realistic or necessarily comfortable for this format. You're more likely to be in a living room, on a sofa or at a kitchen table. However, the same sort of rules apply as

to how the space affects the recording. If there are lots of shiny or flat surfaces, you'll get a lot of echo and reflections in the recording. If you can't acoustically treat the entire space, at least cover surfaces with rugs or throws to deaden the sound a bit.

Positioning the microphones might take a little bit of experimentation, but for spoken word, as a general rule, you don't want the mic too close. This is not a vocal performance; it should be a relaxed situation, with the mic placed where it's not in the way of the conversation. If you are using capacitor microphones then start by placing them 12-18 inches away and just to the side so that you can see your co-host (this position should also helpfully avoid any plosives). Dynamic mics might need to be closer, but try not to threaten the podcasters with them. »



■ If you want to video your podcast for release on platforms such as YouTube, you'll need to make sure the video and audio are in sync. The easiest way to do this is to film and record a handclap and then, with the audio extracted from the Video track, line it up with your recorded audio in the timeline.

HALF THE SIZE ALL THE POWER

avantis **SOLO**

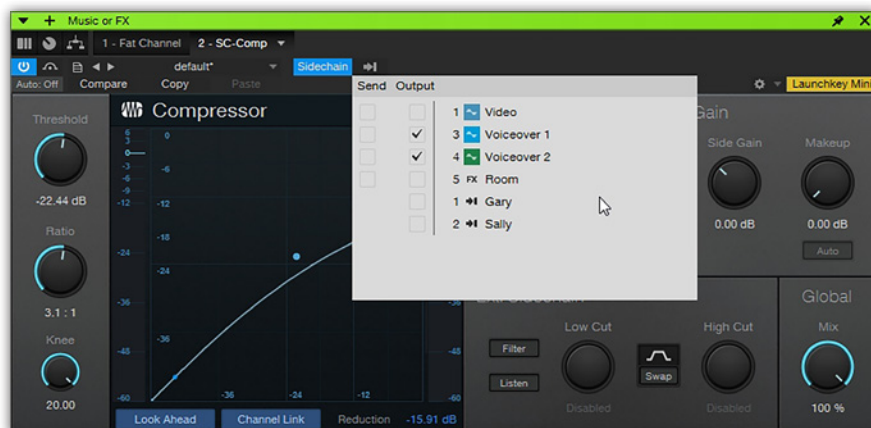


IT'S TIME FOR YOUR **SOLO**



ALLEN & HEATH

www.allen-heath.com/avantis



The music channel in the podcasting template comes pre-loaded with a side-chain compressor. By assigning your vocal tracks to the side-chain input, you can have the music automatically duck down in level when someone is talking.

» Monitoring is easy, because you are literally sat across from your co-host and so you probably won't need any. If you are using pre-recorded music, it would be useful to be able to hear that, but personally, I feel that headphones are just going to get in the way and make you feel less connected to the conversation, so use some speakers.

In Studio One you can check your connections by going to the Song menu and choosing Song Setup. On the Song Setup window, click on Audio I/O Setup and you'll see the inputs from your audio interface listed, along with how they are presented in Studio One. All you need is two mono inputs attached to the two inputs on your audio interface that have the mics plugged into them. While you're in the Audio I/O Setup window, you can rename each connection to something useful like the name of the podcaster. These names will then apply themselves to the channel strips, making it super easy to see which channel applies to whom.

Check the input level by clicking on the Record Enable button and watching the input meter on the left of the channel strips. This will usually also engage the monitoring, which you won't want if you are using speakers, so click the little speaker icon next to the Record Enable button to turn it off. Assuming you're recording at 24-bit, you're looking for a level somewhere between -18 and -10 dBFS. Make sure your podcaster is comfortable and adjust the position of the mic and the gain through your audio interface until you hit the right level.

Processing

The Studio One Podcasting template will have set up the Pro EQ, De-Esser, Fat Channel and Limiter on both voice channels. Let's see what they're doing.

The Pro EQ has a deep low-cut filter of 48dB/oct set to 50Hz, which effectively works to remove any low-frequency rumble or mic handling noise. At the other end there's a high-cut filter dropped in at 20kHz to remove the sort of hiss that no one can hear, but you can always pull that down a bit to be more useful. De-Esser is doing what it's designed to do, in softening any sibilance or overbearing sses. The Fat Channel is not doing a whole lot and is mostly there to add a bit of warmth, but it's ready and available for you to push in a little compression or EQ as you see fit. However, the gate is active at a very low threshold and so shuts down the audio whenever that person isn't speaking. It's a great way to minimise background noise and you can experiment with the threshold to make the best use of it. I would use it sparingly, or it can sound a bit choppy to the listener. The Limiter plug-in at the end of the chain is there to keep a lid on any peaks. While you don't really need to compress the voices much in a podcast, it's good to keep the rise and fall of conversation, and the occasional laugh, under control.

On the music channel we have a Fat Channel with a little bit of compression, EQ and limiting enabled, which are probably unnecessary because your music is likely to already be mastered to a suitable level. However, under that is a side-chain compressor, which might be very useful indeed...

Interacting With Music

Naturally, you are going to want to pepper your podcast with music — that's what the music track is there for. If we are following our minimal-editing, live-like-radio vibe then we're going to want our music to be there while we are recording our

podcast. The great thing is that we can use that to our advantage by placing the audio on the timeline and use it to guide and constrain our podcast.

You'll probably need some intro music, outro music and maybe something incidental between sections. The way the structure tends to go is that your podcast will start with you introducing the name and episode number ("Hi, this is the *Sound On Sound* Studio One Workshop podcast, episode 76"). There's music in the background that then leaps to the fore as soon as you stop speaking for a few seconds of intro music. This then gradually fades out as you begin the podcast for real. For the outro you usually get a slow fade-in before the main outro music kicks in to close your podcast. In the middle you'd often get a short fade into a burst of music to introduce another topic or section. All of these things arranged on the project timeline give you visual and aural clues as to what's about to happen so that you can time your talking to coincide.

Talking can easily get lost in the music, and so you'll need to ensure that when you're speaking the music level is reduced. There are two ways of doing this. You can duck it like on the radio, or you can prepare the audio beforehand.

Duck Duck Go

Remember the side-chain compressor that was dropped onto the music track? Well, that's designed for this exact purpose. It will use the voice tracks to trigger the compressor on the music track, pulling the volume of the music down whenever you speak. The advantage of this method is that you can just let it work its magic without thinking about it, and talk over as much music as you like.

If you open up the SC-Comp on the music channel you'll see that the side-chain button is lit in blue. Click the arrow to the right of it and it will bring up the source selection for the side-chain that will trigger the ducking. If you are trying to record your podcast in a single take live then you'll want to use the input channels. If you are going to do a bit of editing afterwards, which I think is advisable, then you only really need it on playback, so it can be set to the output of the voice channels. You can then adjust the threshold to get the severity of ducking you're after.

Prep School

Ducking isn't to everyone's taste, and it can feel too much like you're running

a pirate radio station. If so, you can do a little bit of audio editing to dictate exactly where you'll be speaking and where you won't.

Take your intro music, slice it up so you have a short segment at the front for the episode number, a longer section at the back where you start the podcast, and a middle that will play as your intro. Use the gain handles on the front clip and pull it down to a good background level. Select a little range between the reduced music and the main intro and hit X to

crossfade between the two. Do the same to crossfade back from the intro music to the background version, and then use the standard gain handles to fade it out. You can see exactly where you have to speak and when you should stop. This is particularly useful for the outro, as you can create a long fade at background levels so you can hear the end of the podcast coming, and know that you have to wrap it up before that crossfade into the outro music. It gives your podcast the same length and rhythm and keeps you focused.

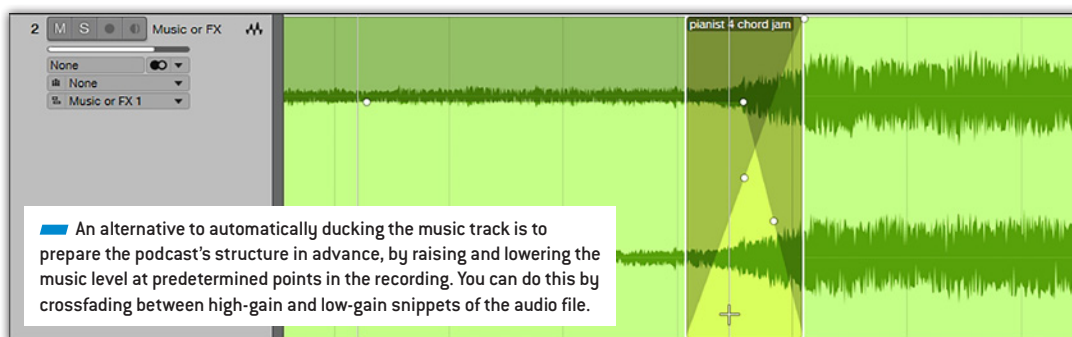
A quick tip is to select the music slices together and hit Ctrl+B (Cmd+B) to generate a mixdown to a single clip so that you don't inadvertently pull the slices and crossfades apart.

Editing

For a natural 'live' feel you won't want to do much editing, but there may be the odd cough or doorbell ring that it would be good to remove. If it's something you noticed during the recording then it's a good idea to drop a handclap in there so you can quickly identify it on the timeline. Also, ensure

that Ripple Editing is enabled so that after each cut the rest of the track automatically moves to fill the gap.

Once you have everything as you like it, save the project as a template, ready to record your next episode! **///**



Drive Your Sound



K R A F T U R

Shape your Power & Enrich your Mix
Loudness Enhancement & Saturation

<sound|theory>

Alchemy's granular engine is a sonic playground with infinite possibilities!

SIMON PATERSON

Who needs a gaming console when sound design offers a world of immersive experiences! If you're ever stuck for a starting point for a new composition, you could do much worse than trying granular resynthesis in Alchemy — it can transform any scrap of audio you have lying around into a goldmine of ideas in no time.

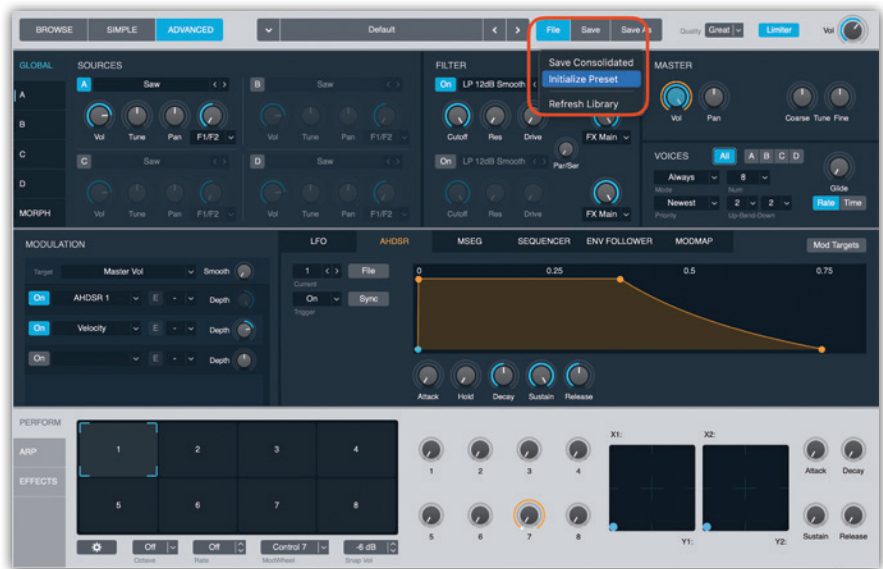
The more you play with Alchemy (I use the word 'play' deliberately), the more you'll find yourself awash with project starting points. You'll also discover endless potential for developing and manipulating your samples.

Composer Matthew Herbert once talked of how we now have the capacity to listen to the world in different ways. I remember being very inspired by his stated intent to "freeze history, press pause, wander around inside the sound". If you are as inspired as much as I was by that idea, you'll relish beginning that journey with Logic Pro's Alchemy synth.

DAWs Of Perception

While Logic's array of virtual analogue synths (ES1, ES2, ES-M, ES-P, ES-E, Retro Synth) can convincingly emulate analogue technology (whilst also offering such digital luxuries as velocity sensitivity and practically unlimited polyphony), Alchemy implements a form of synthesis that can only exist in the digital realm.

Alchemy can function as a powerful standalone synthesizer or as an advanced sampler — which, incidentally, can open any of your EXS Logic Sampler patches. But perhaps it's best to



■ The File / Initialize Preset option will reset Alchemy and pare things down to just Source A — an ideal starting point for your granular experiments.

think of it as a combination of both: a sample-manipulation synthesizer. To begin programming, you start with at least one Source (up to four). These Sources may contain either audio files as starting points or oscillators that will generate all manner of virtual analogue waveforms, from the simple sine through to more complex shapes.

For now, let's focus on granular resynthesis, which, like any time-stretching algorithm, works by breaking up audio into tiny fragments called grains. In Alchemy, these grains can be reordered, shaped, pitch-shifted, repeated, layered, and spread around the stereo field, before being recombined back into an output stream to be further manipulated,

modulated, and effected in a virtually infinite number of ways.

Inner Source

Any sound design session should always begin with a clean slate, so click on the File button at the top of

the plug-in (next to the Preset field) and select Initialise Preset. This resets all the sources, along with their snazzy-sounding modulations and effects. Source A will now become the sole source, and we'll begin by replacing the default Saw wave with a sample of our choice.

Choose anything, but I've found that Alchemy works best with shorter sounds — around 2-10 seconds is ideal. These can be small vocal or monophonic instrumental phrases perhaps, or snippets of field recordings. What's more important, however, is an open mind: expect to open up a rich seam of sonic source material.

To take a snippet out of a larger audio file in the main window, make a rough cut with the marquee tool (I've permanently set it up as my Command-click tool). Swipe to select the part of the audio file you want to process, then single-click to chop it out. Finally, press W to fine-tune the region marker start and end points in the Wave editor.

Ongoing improvements to Logic Pro continue to speed up workflow in meaningful ways; you can now drag and drop your region directly into Alchemy's Source panel. Doing so will provide the four options of sample import: Additive, Granular, Spectral and Sampler. It's crucial to choose the appropriate one at this stage.

Once you've dragged your edited region onto the Granular panel, click the Source A button (on the left) to access all the relevant parameters and delve into the



■ You can now drag audio directly from the arrangement window into Alchemy's granular engine.



Setting up modulation is as easy as right-clicking the destination and picking a source from the list.

stream and is a great candidate for targeting with a source of modulation. For now, simply toggle this up and down, listening to how the grain size affects the quality of sound.

To create a rhythmical pulse, you might have the grain size at between 80 and 230 ms with Density set to 1. Modulating the Size parameter will help to vary the otherwise repetitive rhythm, should you wish to do so and, depending on the size you choose, changing the Grain Shape can have a profound impact on the stream. Try swapping from the default Tukey to Ramp Down, for example, to create a more defined, punchy, rhythmic attack.

For a more sustained, stretched sound, try increasing Density, which changes the number of overlapping grains from one (no overlap) to up to 10.

Num Taps adds up to eight retriggers of the source's attack portion and works in combination with its partner, Tap Spacing. Set to zero, these retriggers stack up on top of each other, but set to a positive value (up to 100) and with the right combination of source file and settings, this combination can create inspiring harmonies that stagger themselves into a kind of 'canon', as if performed in a round.

Next, you might try RTime, which randomly varies the positions of the grain extracts to create a less uniform, more smudgy effect, and finally, try adding some RPan (top right) which adds random panning to nicely fill out the stereo spectrum.

Get In Shape

Naturally, Alchemy offers more than one envelope generator, but the main ADHSR (the 'H' adds Hold to

Here, Pitch is being modulated by LFO 1, which is set to a random 16-step sequence.

the standard set of envelope parameters) has a very intuitive interface for shaping the sound, to easily switch between single, one-shot sounds or more pad-like, lengthy textures, for example.

To create a perfectly playable staccato instrument, set Attack and Sustain to around zero before adjusting the Decay time according to how long you want each note to last. Try 500ms for starters. Then, by adjusting Decay in conjunction with the Speed and Position parameters, you can quickly concoct a unique, playable instrument from virtually any sound source. Simply increase Sustain to create a longer sound.

To add variation, modulation is simple to implement, just right-click any parameter you want to modulate and select the source. Then, look at the Modulation panel on the left and enable the effect by applying some depth. There are so many sources, it would require at least another entire article to even begin to scratch the surface!

Effects

As always, adding effects can help bring any sound to life. While you can use all the effects in Logic as inserts or on busses, Alchemy is great for building a more tightly integrated and flexible effect chain. You can affect each Source separately, or apply effects globally. Click on the Effects tab at the bottom, press On on the first row, and choose an effect from the drop-down menu. Alchemy even includes a version of Space Designer. Finally, if you click on File, you can save (or load) complete effects chains.

I hope you enjoy experimenting with granular resynthesis in Logic Pro. With a little experimentation, improvisation, and an open mind, you're sure to strike gold! **////**



details in Advanced view. The top third of the GUI now shows, from left to right, the Source controls, the waveform display, and what Alchemy calls the Element controls, used to manipulate the grains.

Now for the part where we get to press pause and truly wander around the sound. As you meander around, you'll be transforming the sound in a variety of ways, and depending on your objective, you'll find yourself with more than a soupçon of seductive starting points. Consider a trail-of-breadcrumbs approach: as you encounter sounds you like, save iterative presets that you can revisit later.

Consider Your Position

Before we even explore the advanced element controls, you'll have plenty of fun just playing with the main panel. In addition to the more obvious controls such as pan and tuning, try adjusting the Position parameter while triggering your MIDI keyboard to find your starting point. Like any parameter, this can be targeted not just by standard envelopes, MIDI CCs and LFOs, but also Multi-Stage Envelope Generators, sequencers, arpeggiators and more, to modulate and vary the attack portion of your sound.

Next, there's a Speed control, and this is where you cross the threshold of the event horizon into quantum levels of atomised time and space. Play with just these two parameters and you'll soon open up a parallel universe of possibilities.

Also on the main panel, there are no fewer than seven options for looping, plus you can process your sound with up to three filters.

Grains Of Truth

Now, let's dive into the Granular-specific Element controls. The Size control has significant bearing on the output

Spotlight



SSL Revival 4000

LUKE WOOD

When mixing consoles dominated the studio world, every mic preamp would be followed at the very least by some basic EQ and an insert point to patch in outboard processors. Large-format desks would often include comprehensive filtering and equalisation, and sometimes per-channel dynamics processing too, and it was perfectly normal to use these facilities while recording, to get the sound right before it made it on to tape. Today, many studios are entirely console-free, relying either on rackmount preamps or those built into their audio interface, and many engineers tend to look to the DAW and plug-ins, rather than outboard, for processing. Thus, other than perhaps applying the high-pass filter on a mic or preamp, it's common not to process signals at all until after they've been recorded.

But there are still some solid arguments in favour of the 'old' approach. Making decisions and committing to sounds while recording can build confidence and generally leaves you with far less work when mixing. Many engineers say they can get a better sound out of hardware, of course, but many others simply favour the hands-on approach and screen-free immediacy of interacting with physical gear.

Hardware Channel Strips

We're decades into the age of the DAW, but outboard channel strips remain popular for good reasons — and there are plenty to choose from.

So, a channel strip that combines your mic preamp with a handful of other processors can be a genuinely useful addition to any studio, providing a 'front end' to your DAW, and potentially also a processing strip that you can access when mixing.

As with most kit, there's an overwhelming number to choose from. At the budget end of the market you can find fairly feature-rich options from the likes of **PreSonus** and **dbx**, while for just a little more you have the likes of the **ART Pro Channel III** that's reviewed in this issue. If you take a no-expense-spared approach, you might prefer to check out **HUM Audio's N-Closure** system (which houses the comprehensive channel modules from their high-end **N-Trophy** console), and **elysia's** recently announced **Channex Studio**. There are more traditional high-end valve-based options from the likes of **Avalon**, **Chandler Limited** and **Manley Labs**, as well as cleaner-sounding high-quality preamp and EQ strips from companies like **George Massenburg Labs**.

Between those extremes are all manner of devices, offering different sounds and facilities ranging from basic EQ and compression to de-essers, saturation processors and even transient shaping. Units such as **Gainlab Audio's Bishop** aim to replicate the sound and functionality of vintage valve gear for a relatively keen price, while some take a more modern approach and focus on delivering a combination of transparent preamplification and processing — **DAV Electronics' BG5 MkII**, for example. Others take a best-of-both-worlds approach. Given that so many studio users are now working without a mixer, it's not surprising to see that the designers behind some of the most iconic large-format beasts, such as **SSL**, **API**, **Harrison**, **Neve** and **Rupert Neve Designs**, now offer rackmount units based on their most sought-after console channels. And with all that in mind, let's take a closer look at a selection of the available options.



API TCS-II

API TCS-II

The TCS-II brings together a selection of API's famed circuits to offer a compact and relatively affordable way to access the sound of their much-loved consoles. Preamp duties are taken care of by the 512C, which will accept mic, line or instrument-level signals, and offers up to 68dB of gain. The 527A compressor follows, with threshold, attack, release and ratio controls presented on detented pots, and API's patented Thrust side-chain circuit offering the 'in-your-face' compression style for which the design is so well known. When it comes to EQ, the TCS-II employs the 550A design, a three-band circuit with selectable frequencies, a choice of shelf or bell filters for the high and low bands, a switchable 50Hz-15kHz band-pass filter, and ± 12 dB of gain for each band. Finally, the output stage is based around the 325 Line Driver found in the company's consoles. A front-panel switch allows the compressor to be placed pre- or post-EQ, and as well as being individually bypassable, each of the unit's sections can be accessed separately thanks to a collection of rear-panel I/O sockets.

\$ \$3495.

W www.apiaudio.com

ART Pro Audio Pro Channel III

The third iteration of ART's recording channel boasts an impressive combination of features for such a keenly priced unit, and manages to sound better than that price might suggest. As well as a clean-sounding preamp stage with a fully variable high-pass filter, there are separate insert send and return jacks for patching in other processors, an intuitive four-band EQ comprising a high and low shelf and two swept mid bands, and, unusually, a dual-voice compressor that gives you the option of valve-amplified optical compression or a FET compressor that you can set to be more assertive.

\$ \$499.99.

W www.artproaudio.com



ART Pro Audio Pro Channel III

Avalon VT-737SP

The preamp section of Avalon's VT-737SP employs a pair of dual-triode valves and features independent mic, line and instrument inputs placed in front of a variable high-pass filter. The following section houses an optical compressor that's capable of everything from gentle compression to hard-knee limiting, and offers plenty of control thanks to threshold, ratio, attack and release knobs. Next up, a four-band EQ boasts a discrete Class-A signal path, and low and high shelving bands that offer the smooth character associated with passive designs, while two mid bands benefit from variable frequency and Q controls, along with x10 multiplier switches that extend their frequency range. A handy option is the SC Mids switch, which allows the two midrange EQ bands to be placed into the compressor's side-chain to finesse its response, while the low and high shelves remain in the main signal path. The output stage utilises yet another dual-triode valve.

\$ \$3866.

W www.avalondesign.com

Chandler Limited TG12411

Chandler Limited's Abbey Road Series comprises an array of outboard processors that are officially licensed by the legendary London studio. The TG12411 combines a version of their TG2 preamp with a passive EQ that's based on the circuit found in the TG12410 Transfer Console — that's the mastering counterpart to the famous Beatles console. Mic, line and instrument-level signals are supported, and the gain control is followed by an output level knob, meaning that the preamp section can be deliberately driven for effect without the risk of clipping downstream devices. The EQ section features a capacitor-based high-shelf band, followed by an inductor-based Mid Peak band with proportional Q (there's a toggle switch for narrowing the overall curve), an inductor/capacitor Mid Cut band with a default narrow Q, and an inductor-based Low Boost with proportional Q. Finally, a Low Cut switch offers a choice of five frequencies. Boosts and cuts of up to 20dB mean the unit can apply some pretty heavy-handed curves, and the independent cut/boost bands mean that the TG12411 can behave in a similar way to Pultec and Lang EQs.

\$ \$2600.

W www.chandlerlimited.com

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Avalon VT-737SP

Chandler TG12411





Drawmer Electronics 1977

» Drawmer Electronics 1977

Said to be their most comprehensive channel strip to date, Drawmer's 1977 is certainly rich in features. It places a clean mic pre in front of a three-band EQ and soft-knee compressor to provide the classic channel strip options, but this design also boasts three styles of variable saturation. A stepped control delivers up to 64dB of gain for mic, line and instrument sources, while a choice of three impedances ensures compatibility with a wide range of microphones, as well as the ability to coax different voicings out of passive dynamic models. The three-band EQ offers ± 12 dB of adjustment per band with adjustable Q for the mid band, as well as a selection of filter-slope options for the low- and high-shelf bands. A compressor equipped with threshold, ratio, attack and release pots follows, and offers further tonal options courtesy of Drawmer's signature Big and Air modes. Yet more tonal shaping is provided by a variable Saturate control, which offers a choice of three styles: clean, warm and overdriven. Helpfully, the compressor can be placed before or after the EQ, and the unit's rear panel offers a clean direct output from the preamp section, in addition to its main output at the end of the chain — so you can, if you wish, record 'clean' while monitoring with the processing, or record both clean and processed versions for options at mixdown.

\$ \$1979.

W www.drawmer.com

Grace Design m103

Grace Design products are renowned for their impressive technical performance and clean sound, and the m103 is no exception. Preamp duties fall to the company's m101 design, which accepts mic, line and instrument sources, and features a Ribbon mode that both increases the input impedance and blocks phantom power. A three-band EQ offers a choice of peak or



Grace Design m103

shelf filters for the low and high bands, along with a fully parametric mid. This is followed by an optical compressor with independent user control of the threshold, attack, release and ratio. Finally, a Trim dial provides ± 10 dB of level adjustment at the output stage. The order of the compressor and EQ can be selected using a front-panel switch, and additional rear-panel connections make it possible to patch the preamp and EQ/compressor sections separately.

\$ \$2358.

W www.gracedesign.com

Great River Electronics MEQ-1NV

Great River Electronics' MEQ-1NV is technically two products (the half-rack ME-1NV preamp and EQ-1NV EQ), but since it's possible to order them as a pair assembled in a 1U rackmount frame, we felt it should be included. The compound device accepts mic, line and instrument signals and delivers up to 70dB of gain, while custom-wound Sowter transformers deliver a vintage-inspired sound. Four EQ bands with ± 15 dB of gain follow, with the low and high bands offering a choice of peak or shelf filters, and the two mid bands being equipped with three selectable Q values. There's also an 18dB/octave high-pass filter that can be placed at six different frequencies.

\$ \$3575.

W www.greweb.com

Harrison 32Classic MS

Another channel strip derived from a large-format console is Harrison Audio's 32Classic MS, based on their recent revamp of the iconic 32C. There are dedicated rear-panel mic and line inputs, along with a handy front-panel combo socket that'll accept mic, line or instrument-level signals, and independent gain controls are provided for mic and line gain. As you'd expect, the 32Classic MS features Harrison's signature filters: a 25Hz-3.1kHz high-pass with a 3dB resonant peak, and a 160Hz-20kHz low-pass. In addition to the filters there are four bands of EQ, each offering ± 10 dB of gain, selectable frequencies and a proportional Q behaviour. Although there's no dynamics processing, the 32Classic MS does feature an insert point, so you can place external processors in the signal path, and if you're keen to keep things authentic, Harrison's own console-derived compressor is available as a 500-series module. The right side of the front panel is dedicated to a Mixbus section that includes an output 'fader' and pan controls, and allows multiple units to be linked using rear-panel connections and their signals summed to a single stereo output.

\$ \$1999.

W www.harrisonaudio.com

»



Great River Electronics MEQ-1NV



Harrison 32Classic MS

Mix and Match

Customize your API 2448 to match your passion and workflow, with a wide variety of configurations available.



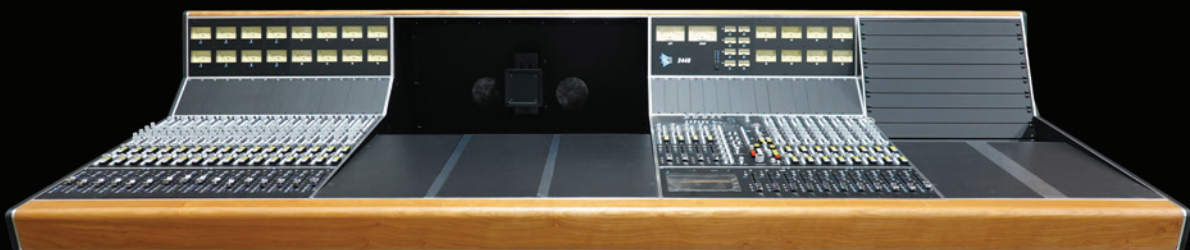
24 Channel API 2448 Console



With 30" Producer's Desk



With Dual 19" Racks



With 30" Producer's Desk, and 19" Rack



apiaudio.com



Heritage Audio BritStrip

» Heritage Audio BritStrip

The BritStrip brings together Heritage Audio's 1073-inspired preamp and EQ offerings, and throws in the diode-bridge compressor from their Successor unit for good measure. There are custom Carnhill input and output transformers, and the preamp offers up to 80dB of gain, along with two switchable input impedances. As well as accepting mic and line-level signals, the preamp features a JFET instrument input with a DI-style thru output for passing instrument signals on to an amplifier or pedalboard. A three-band EQ follows, with a trio of switches allowing the high and low shelves to be switched to peak filters and engaging a narrower Q setting for the mid band. The compressor has threshold, make-up gain, ratio and release controls, along with a Fast Attack switch, a selection of five side-chain filter frequencies and a Blend control that allows for instant parallel compression. External side-chain insert points are also provided.

\$ \$1999.

W www.heritageaudio.com

Manley VoxBox & Core Reference

Designed to be the ultimate all-in-one toolkit for vocal recordings, the VoxBox combines Manley's valve-based Mono Mic Preamp with a selection of processors that draw on their other high-end gear. Mic, line and instrument-level signals are accepted, and the preamp's gain switch also affects the circuit's slew rate, so as well as defining the input signal level it affects the tonality of the unit somewhat. The compressor employs an optical design based on the company's ELOP limiter, and offers a fixed 3:1 ratio along with threshold, attack and release knobs. Despite their familiar-sounding names, the controls



Manley VoxBox

work slightly differently from those on a typical compressor, applying up to four different time constants simultaneously and delivering a wide range of compression characteristics thanks to a technique that Manley have dubbed 'parametric compression'. A Pultec-inspired EQ includes Low Peak, Mid Dip and Hi Peak controls, each with a generous set of frequency options, and there's also a De-Ess section that can be targeted at the 3, 6, 9 or 12 kHz range, or set to act as a 10:1 limiter designed to mimic the sound of an LA-2A. Despite its name, Manley say that the VoxBox also excels on instrument-miking duties — but if the VoxBox is overkill, they also offer the more affordable Core Reference. This model employs the same preamp topology as the VoxBox, but pairs it with slightly simpler compressor and EQ sections, and includes a dedicated limiter.

\$ VoxBox \$5499, Core Reference \$2899.

W www.manley.com

Neve 8801

Neve's 8801 houses a complete channel strip from the company's acclaimed 88RS console in a 1U rackmount chassis. The Marinair-transformer-equipped preamp is followed by independent low- and high-pass filters, and a dynamics section that features a VCA compressor/limiter and a gate/expander. Threshold, make-up gain, ratio and release controls are provided for the former, and two switchable attack times

are available, while the latter is equipped with threshold, range and release controls, along with a Hysteresis knob that defines a separate threshold for rising and falling signal levels. Next up is the EQ section, which includes low and high bands that can be switched between bell and shelf modes, and a pair of fully parametric mid bands. Helpfully, the 8801's filters and EQ section can each optionally be inserted into the compressor's side-chain circuit, and an Audio Routing switch makes it possible to place the unit's dynamics, EQ and external insert points in any order you wish. The 8801 can also be fitted with an optional A-D card that provides S/PDIF, AES3 and DSD digital outputs, in addition to the standard analogue I/O.

\$ \$3695.

W www.ams-neve.com

Rupert Neve Designs Shelford & Newton Channels

Rupert Neve Designs' flagship Shelford Channel serves as a 'best of' collection, bringing together some of the legendary designer's most praised designs. Its transformer-coupled preamp delivers up to 72dB of gain, and is joined by an instrument input derived from the company's RNDI range plus a variable high-pass filter. The inductor-based EQ design features peak/shelf low and high bands, with the former inspired by the 1064 and the latter offering a hybrid vintage/



Neve 8801



Rupert Neve Designs Sheldford Channel

modern sound thanks to a mixture of inductor and capacitor-based circuitry. The mid band is based on the 1073, and is capable of everything from subtle enhancement to surgical correction thanks to a proportional Q design. Next up is RND's diode-bridge compressor, a design renowned for its up-front character. There's a wealth of control on offer thanks to threshold, ratio, gain, timing and blend controls, and it's possible to place the unit's high-pass filter in the side-chain. Lastly, switchable Red/Blue Silk circuits make it possible to introduce a generous dose of harmonic distortion. A more accessible entry into the RND world comes in the form of the Newton Channel — you still get a high-quality preamp section and the Red/Blue Silk circuitry, but it lacks the RNDI instrument input, and it employs a simplified EQ section and cleaner-sounding VCA compressor.

\$ Sheldford Channel \$3999,
Newton Channel \$1999.
W www.rupertneve.com

SPL Channel One & Track One Mk3

SPL's flagship Channel One Mk3 packs in a comprehensive range of processing, and includes some particularly versatile I/O options. The preamp delivers up to 68dB of gain and caters to all the usual signal types, but unusually it includes two separate mic inputs to provide a convenient way to carry out A/B comparisons. A Tube Saturation control

then delivers access to variable harmonic distortion, while a Tube Post switch places the circuit either after the preamp or in between the EQ and output stages, to give even more tonal flexibility. The signal then passes through De-esser and Transient Designer sections, before reaching a simple compressor equipped with compression and make-up gain controls. As for EQ, the unit includes a pair of midrange bands with ± 12 dB of gain, and an Air band that takes the form of a bell filter centred at 19kHz. The Track One features a console-style direct output from the preamp section, and parallel main outputs that allow it to feed two or even three destinations. SPL also offer the Track One Mk3, which is more compact and omits the Tube Saturation and Transient Designer sections, and the additional I/O.

\$ Channel One Mk3 \$2999,
Track One Mk3 \$1199.
W www.spl.audio

SSL Revival 4000 & Super 9000

As its name suggests, the Revival 4000 brings the sound of SSL's iconic 4000-series consoles to their current rackmount product line-up. Its transformer-balanced preamp employs the same components found in the original E Series design, and is followed by a de-esser taken from the B Series variant. The following dynamics section includes a VCA compressor and gate that stay true to the E Series circuitry, with the compressor offering a versatile range of characters thanks to a soft-knee bypass, a fast attack

mode and the ability to switch between logarithmic and linear release curves. Variable low- and high-pass filters follow, and can be placed either in the main signal path or inserted into the compressor's side-chain. The four-band EQ section takes a two-for-one approach, providing both the 'brown knob' variant found in earlier consoles, and the 'black knob' version developed alongside George Martin for AIR Studios' first SSL console. The latter design features an extended gain range (± 18 rather than ± 15 dB), a more refined control response, and a steeper 18dB/octave high-pass filter. Routing buttons allow the de-esser and dynamics sections to be placed post-EQ, and the Revival 4000 also includes a balanced insert point and an external side-chain input. Just last month, SSL announced the launch of the Super 9000, a second channel strip, this time based on the SuperAnalogue circuitry introduced in SSL's later 9000-series consoles, which offered a cleaner, smoother-sounding alternative to the more punchy, mid-forward character of the 4000-series.

\$ \$1999.
W www.solidstatellogic.com

Universal Audio 6176

Universal Audio's 6176 combines a preamp derived from Bill Putnam's 610 modular console with one of the most popular compressors of all time: the 1176LN. The all-valve preamp section will accept mic, line and instrument-level sources — two impedance options are provided for the mic and Hi-Z inputs — and is paired with low- and high-shelf EQ bands that each feature ± 9 dB of gain and three corner frequency options. The unit's right-hand section houses the 1176's attack, release, input, output and ratio controls, the last of which includes a setting for the infamous 'all buttons in' mode. A front-panel toggle switch provides Join and Split operation modes, with the former placing the preamp and compressor in series and the latter allowing them to be patched in separately via individual rear-panel connections.

\$ \$2999.
W www.uaudio.com



SPL Channel One Mk3



Universal Audio 6176

40 IDEAS THAT CHANGED MUSIC

Our four-part series charts the most important technical breakthroughs in music history.

THE SOS TEAM

A lot has happened since the first issue of *Sound On Sound* appeared in October 1985. Music technology and the music industry have changed beyond all recognition. Today's home computers can do things that would have seemed impossible in the mid-'80s. But how did we get here? To mark *Sound On Sound's* 40th anniversary, we have put together a list of 40 milestone developments from the history of recording and production, which we'll be counting down over the next four issues. Agree? Disagree? What have we missed? Get in touch on feedback@soundonsound.com to share your own thoughts.

40: Spectral editing

Spectral editing is a digital-audio processing technique that allows users to view and manipulate sound using a spectrogram display that simultaneously shows time, frequency and amplitude, rather than just a waveform. First seen in CEDAR Audio's Retouch process, launched in 2002, the display makes it possible to visually identify and isolate different sounds that occur simultaneously, thereby facilitating the removal of clicks, hums and other background noises often without affecting the surrounding audio. Retouch, along with Steinberg's SpectralLayers (originally developed by Divide Frame and published by Sony) and iZotope's RX software, has a signal-focused approach optimised for use in post-production and restoration, whereas a more music-based app like RipX uses a slightly different spectral analysis that is more note- and harmonics-based, with a focus on instrument separation. Melodyne, similarly, uses a form of spectral note extraction alongside polyphonic pitch detection to create its unique visual editing interface.



— Producer Roger Nichols developed the first drum sample replacement system.

39: Drum sample replacement

Steely Dan producer Roger Nichols bought his first computer in 1976. By 1978 he had developed the first version of Wendel, a drum machine that employed 12-bit, 125kHz sampling. It wasn't the first digital drum machine but, uniquely, it had an envelope follower that allowed it to be triggered from a pre-recorded drum track. Existing kick and snare sounds could be replaced with factory samples, or you could sample and replay clean hits from your own multitrack. Wendel was first used in January 1979 on Steely Dan's album *Gaucho*, and was superseded by the 16-bit MkII version in 1981.

38: Parallel compression

Unlike standard compression, which reduces peak levels with the aim of controlling overall dynamic range, parallel compression blends a heavily compressed version of an audio signal with the original unprocessed signal. This has the effect of maintaining all the transients and dynamics of the peaks whilst still adding density through raising the level of quieter parts of the source via the compressed path. The most common setup is to route a track to a secondary channel or auxiliary bus, where heavy compression is applied,

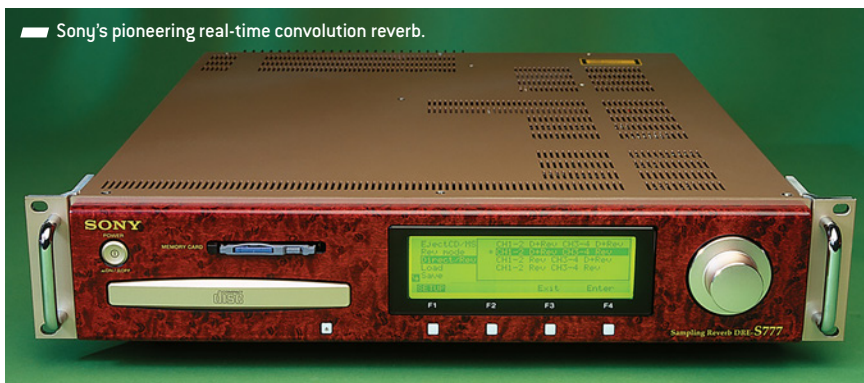
before mixing that signal alongside the dry track. The audible effect is denser and more powerful, just like normal compression with some make-up gain applied, but without any loss of transient detail — a common side-effect of compression — because the dry, uncompressed track is always louder on the peaks. It's commonly used on drums, sometimes vocals, and even full mixes to increase average loudness and perceived energy. The technique is sometimes referred to as 'New York compression' because of its frequent use on New York disco and pop recordings in the 1970s and 1980s.

37: Re-amping

Re-amping, as used in the context of guitar recording, refers to recording a clean, direct signal via a DI box that can subsequently be replayed through a guitar amp and re-recorded. The player will usually be listening to a guitar amp at the time to help with the original performance, but the technique allows the sound to be altered or improved later without requiring the part to be played again. When re-amping it is important for the playback signal to match both the level and source impedance of a typical guitar signal to ensure that the amplifier will react normally in terms of distortion and tonality. A number of dedicated impedance-matching boxes are now available, notably Radial's X-Amp. Re-amping is often cited as being 'invented' in the early '90s, but was actually in regular use well before, using either custom-made adaptors, or even passive DI boxes connected backwards, so the amp would see the high-impedance side of the box.

36: Noise reduction

Dolby Laboratories developed their A-type process in 1965 to combat the high-frequency hiss inherent in analogue tape recording. In the following years, as multitrack tape recorders added more



■ Sony's pioneering real-time convolution reverb.

discrete tracks to the standard tape widths, the Dolby process became an essential component of many professional multitrack recording studios. The rival dbx system, using a full-bandwidth compander to compress the signal during recording and expand it on replay, was effective at suppressing noise, but could also produce audible noise modulation. The Dolby system avoided this by dividing the audio spectrum into four separate frequency ranges, applying compression and complementary expansion separately in each band, reducing noise by up to 10dB in the midrange and 15dB at high frequencies without significant side-effects. Dolby-based systems always had to be carefully calibrated to ensure that the record and replay levels matched, otherwise the encoded signal would not be properly 'decoded'. By the late 1980s, higher fluxivity tape formulations and 30-ips working were often preferred to using Dolbys for noise reduction.

35: Mixing in the box

Ricky Martin's 1998 global chart-topper 'Livin' La Vida Loca' is notable for more than just its singer's chiselled good looks, or its unusual success in bringing Latin music to a mainstream audience. It's generally regarded as the first number one single to be mixed entirely in software, without the assistance of a hardware console or any outboard processing. The software in question was, of course, augmented by hardware DSP cards, but even so, 'Livin' La Vida Loca' was a recognisable landmark along the path to the current situation where composers and producers can work anywhere, with only a laptop and a pair of headphones.

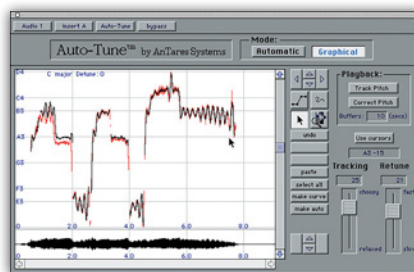
34: Convolution

Convolution exploits the idea that every single sample in a digital recording can be considered, in its own right, an instantaneous impulse with a given amplitude. So if we

have another digital recording that actually represents the response of a room or other structure to an impulse, we can impose the sound of that room on the first recording by playing back the second one over and over again, with its amplitude adjusted to that of each sample in the first. This process was understood many years before it became practical to implement, with the first real-time convolution reverb being Sony's DRE S777 from 1999. Not all of the claims made for convolution in its early days were fulfilled, because the realism of simple implementations is compromised by its inability to capture time-variant or non-linear phenomena, but many of these shortcomings have now been addressed. As well as powering many reverb plug-ins, convolution is now also the standard technique behind loudspeaker emulation in amp simulators.

33: Pitch correction

In 1996, physicist Andy Hildebrand was inspired to apply some of the insights he'd gained in the oil and gas industry, where he worked on improving the interpretation of radar images, to audio. The result was something previously thought impractical: a real-time algorithm that could analyse and automatically correct the pitch of monophonic audio. This algorithm formed the basis of what may be the single most influential plug-in of all time. Used with discretion, Auto-Tune could eliminate 'pitchiness' from a sung



■ An early iteration of Antares' Auto-Tune plug-in.

vocal in a remarkably transparent way. But what no-one had anticipated was that producers would use it without discretion. Cher's 1999 mega-hit 'Believe' showcased the uncanny sound that could be achieved with fast retune speeds; at the time, it seemed like a novelty, but the effect has become part of the standard sound palette of hip-hop and R&B.

32: Time-stretching

The ability to play back recorded audio faster or slower without changing its pitch might seem like a quintessentially digital phenomenon. Not so. German inventor Anton M Springer published a paper in 1957 describing his all-analogue, tape-based Acoustical Time Regulator, which he had demonstrated in operation four years earlier. Four playback heads were mounted on a cylinder which could be rotated independently of the tape speed. Switching between the heads allowed short segments of the audio to be repeated or omitted, depending on whether the tempo was to be reduced or increased. Springer claimed that tempo could be varied by ± 30 percent with no loss of audio quality.

31: MIDI recording sync'ed to tape

From the perspective of today's 'in-the-box' DAW production techniques, it is easy to overlook the mid-'80s technology revolution that came before. Early MIDI sequencers, whether in software or hardware, could start in sync with a multitrack tape machine and continue running in parallel using a rudimentary sync code, but they always had to start from the beginning. Clearly, this was of no practical use in an overdub session where you might shuttle about endlessly. Everything changed, however, with the development of the ability to lock sequencers to industry-standard SMPTE timecode. This meant that the sequencer would start playing in sync at whatever point you started the tape, effectively adding 'virtual tracks' to any size of multitrack setup. In the era before audio recording on computers, software sequencers such as Steinberg's Pro 24 and C-Lab's Notator became perfect companions to the narrow-gauge multitracks of the day, as fundamental elements like drums and bass parts never needed to be recorded to tape at all. It was also possible to edit the MIDI elements of a performance, or indeed change sounds, right up to printing a final mix. ■■■



GEAR OF THE YEAR!

2025

The SOS team pick their highlights from recent hardware and software launches.



AEA 1029

Studio Hardware

At first glance I thought that the **AEA 1029's** compression options might be slightly limited. However, once I'd explored the extent of its capabilities, the 1029 proved to be an impressive performer. It is more than capable of delivering superb, open and transparent audio performance and impressively precise control of peaks across a wide range of sources. The fixed attack and release timings have been very carefully chosen, which made working with the 1029 a hugely enjoyable and intuitive exploration of its possibilities. Overall, the AEA 1029 is a compressor that gets the job done and is, for my needs, the ideal companion for recording with stereo microphone setups.

Bob Thomas

<https://aearibbonmics.com/aea-products/1029/>

The **Tonograf EQ** by **Atomic Analog** is a part-for-part recreation of the legendary TAB W95 equaliser module from the early 1960s — a fast-moving period in

audio design that saw the original designer take the ideas used in earlier tube devices such as the V72 and V76 amplifiers and adapt them to low-voltage transistor circuitry. With simple high and low shelving filters and an inductive mid-band option, these units have a pleasing inherent sound combined with a simple-to-use EQ that works superbly for all-round recording and mixing. It took a huge amount of financial discipline to take them out of my rack and send them back after the review period!

These classy-looking EQs definitely sit in the more luxury end of the market price-wise, but pack an awful lot of analogue muscle into a 500-series module. *Neil Rogers*

<https://sosm.ag/AtomicTonograf>

Although notoriously difficult to engineer well and historically expensive to implement, PWM technology has long been the 'holy grail' of analogue compression, offering the very best technical performance and widest range of artistic control. So **Cranborne Audio** are



Cranborne Audio
Brick Lane 500

to be applauded for delivering a brilliantly well-designed and incredibly versatile PWM compressor, genuinely pushing the envelope of what a compressor can achieve — and at an impressively affordable price, too. The **Brick Lane 500** has been the most

inspiring product of 2025 for me, without a doubt. *Hugh Robjohns*

<https://sosm.ag/BrickLane>

Several tools for seamlessly integrating analogue processing into a DAW environment have appeared in recent years, and **Freqport's FreqInOut FO1** is one of the neatest I've come across. To your computer, it's a USB interface with line-level I/O that can be used to loop in outboard. But to the user, it's a real-time plug-in that can be stored and recalled with DAW sessions. *Sam Inglis*

<https://sosm.ag/FO1>



Atomic Analog
Tonograf



Freqport FreqInOut FO1

Garbage Ideas' debut, the **FK Comp**, was one of the products I most enjoyed playing with in the last year. A relatively affordable, simple-to-use FET compressor, the FK Comp employs a non-linear amplifier stage that encourages you to lean into the more characterful side of compression. With simple input, output and mix controls and an LED light that proudly proclaims "You did it" when the compression threshold is breached, you probably get the idea that this isn't a tool for audiobook recordings — although it is capable of clean level control if used carefully. I loved it for driving drums or vocals into that heavy saturated compression sound, which — thanks to the blend control and the clever design of the circuit — still manages to retain enough clarity to be usable come mix time.

Neil Rogers

<https://sosm.ag/FKComp>

The **Hazelrigg DNE** is a superb compressor that had me reaching for my wallet almost as I switched it on. Its pulse width-modulated, twin side-chain circuits come directly from DW Fearn's stellar VT7 PWM compressor. Each side-chain has its own attack and release time constants, which are blended together to control the DNE's pulse width-modulated gain cell, allowing the compression to be tailored to deliver subtle control of the dynamics of vocals, acoustic guitar, double bass, piano and percussion instruments. Even when larger amounts of gain reduction are needed, the DNE's low ratio, soft-knee style of compression remains incredibly transparent. The Hazelrigg DNE is one of, if not the, finest 500-series compressor that I've had the pleasure of reviewing. I still miss it, and my wallet still worries about that. Bob Thomas

<https://sosm.ag/DSeries>

The **Neve 88C** compact dual-channel VCA compressor borrows heavily from the 88RS console's channel dynamics, including clever features

■ Neve 88C



■ Garbage Ideas
FK Comp



■ Hazelrigg
Industries DNE

such as the adaptive attack. It's all powered over USB-C and this, along with the rugged leatherette chassis, makes it way more portable than most analogue compressors. And Neve have made great use of the front-panel real estate too, with dual-purpose controls, channel linking options, and meters changing colour to denote different modes, amongst other things. Most importantly of all, it's really easy to dial in, and to my ears, sounds smooth and natural

on pretty much any source. Matt Houghton
<https://sosm.ag/Neve88C>

While the **SSL Oracle** wasn't quite the most expensive new mixing console that I laid hands and eyes on this year, it's very definitely aimed at the high-end professional market. And while that sort of money can buy other analogue consoles with recall systems and digital control, there's nothing else that delivers the concept so completely: it's an SSL console with an entirely analogue signal path, but whose every setting can be recalled accurately in less time than it takes to boot your DAW. There's an EQ on every channel that can be set to SSL's E or G-series 'flavour' and the ability to set up sessions remotely. Best of all, to my



■ Tascam Studio Bridge

mind, is that, unlike typical digital mixers, it can offer almost the same 'one-knob-per-parameter' immediacy as a traditional analogue console if you want that. Oh, and it sounds great too!

Matt Houghton

<https://sosm.ag/Oracle>

Sometimes, I don't like screens — they encourage me to use my eyes before my ears — and the **Tascam Studio Bridge** has let me escape them; it's made it practical to put my mixing console back at the heart of recording and jamming sessions. It's the first 24-track recorder for decades that lets you individually arm its 24 tracks, and record overdubs and punch-ins. I can load backing tracks, sketch out a song structure in advance and, when I'm done, transfer the WAV files to my DAW. Oh, and it doubles up as about the best value USB audio interface with 24 line-level analogue I/O there is!

Matt Houghton

<https://sosm.ag/TascamStudioBridge>

»



■ SSL Oracle



ADAM Audio D3V



Amphion One18X



AIAIAI TMA-2
Studio Wireless



Trinnov Nova

» Monitoring

Before I'd reviewed **ADAM Audio's** compact **D3V** monitors, I would never have considered installing a pair of dedicated active computer loudspeakers costing almost £300 on my desktop. However, once I'd heard the results of connecting my computer over USB-C to the D3V's 200W RMS of stereo amplification via its onboard DAC, the ultimate decision was never in doubt. The combination of that amplifier power with the D3V's all-new D-ART ribbon tweeter, 3.5-inch woofers, and twin 3.5-inch passive radiators, delivers a superbly detailed overall performance. Thanks to its passive radiators, the power and accuracy of the D3V's low-frequency extension and delivery belies its compact dimensions, while its wallet-friendly price point only adds to its attraction. *Bob Thomas*
<https://sosm.ag/D3V>

I'm perfectly used to using wireless headphones for gaming and other media, but it was still a strange experience to walk around untethered while listening to my DAW on **AIAIAI's TMA-2 Studio Wireless** cans. This latest iteration of their wireless studio headphone technology, while not lossless like the earlier Wireless+ model, is still easily good enough in quality for most studio purposes (if not mastering), and the latency is considerably lower (certainly low enough for tracking virtual instruments). They're perfect for that 'pacing around restlessly' stage of production, and they're also, like all of AIAIAI's products, commendably eco-conscious in terms of manufacture and repairability. *Chris Korff*
<https://sosm.ag/TMA2Studio>

One of the reasons I still find monitors so fascinating is that speaker engineers continue to find so many different

ways of skinning the cat. There's the contemporary, high-tech DSP route and there's the more traditional analogue artisanal route — and countless options in between. And even if a manufacturer takes the traditional analogue route, there's always improvements and more performance to be found. I reviewed the original **Amphion One18** a decade ago, and even though the 2025 **One18X** version appears to be very similar, the reality is that it seems to have taken quite a significant step up in performance. The One18X demonstrates strong correlation between serious, engineering-based electro-acoustic development and subjective performance. The reduced distortion of the new tweeter seems genuinely audible and has lifted the performance of the entire system. The One18X was a genuine pleasure to have for a while in the studio. *Phil Ward*
<https://sosm.ag/AmphionOne18X>

Room and monitor optimisation systems are not without issues. The primary challenges are the distortion of nearfield monitor frequency response in the interests of 'flat' in-room response, and the general lack of any skin in the time-domain game. The **Trinnov** system has for some time been probably the most effective in terms of managing and minimising the inherent issues, but until its **Nova** system came along, the entry price was high and the user interface was idiosyncratic at best. Trinnov's Nova changed all that. It brought the price down from eye-watering to potentially affordable, and it offers an interface that's easily usable by mere mortals. Of all the review subjects that had temporary studio residence in 2025, Trinnov's Nova is probably the one I'd most wish didn't have to leave. *Phil Ward*
<https://sosm.ag/TrinnovNova>

Microphones

In the world of classical recording, Acoustic Pressure Equalisation (APE) of small-diaphragm omnidirectional capsules has long been a very desirable thing — ever since Neumann introduced the M50 in 1951, in fact — enhancing the HF directivity to improve stereo imaging in spaced mic arrays like the Decca Tree. Many SDC omni microphone manufacturers offer accessory APEs to fit their mics, but **Braingasm's Ocula** is the only model (other than M50 tributes) to feature a permanently integrated APE, which makes it very special indeed. It also happens to be quite attractively priced in comparison to the alternatives, making it a very desirable option for classical recording aficionados. *Hugh Robjohns*
<https://sosm.ag/Ocula>

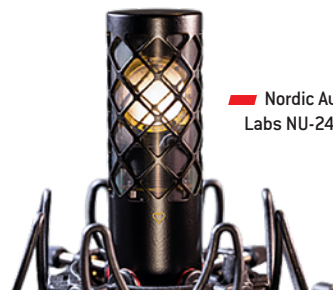
The **Lauten LT-386 Eden** valve microphone was a real feature item in my studio during the month or so I had it for review this Summer, sitting almost permanently in position, ready for the next vocalist, guitarist or drummer to cross its path. This is not a cheap mic by any means, but it does offer a surprising amount of versatility thanks to some clever 'multi-voice' circuitry that changes the mic's signal path. The Neutral setting offers a clean U47-style sound, or you can go a bit darker and more obviously coloured with the Gentle setting. Most striking, though, is the Forward option, which puts you clearly in Sony C-800G territory with a bright, modern vocal sound. There is also a two-stage high-pass filter that adds even more flavour control to this substantial valve mic. One for the fantasy wish list, perhaps, due to the pricing, but well worth checking out if you're in the market for one special mic to do a lot of jobs well. *Neil Rogers*
<https://sosm.ag/Eden>



— Braingasm Ocula



— Lauten Audio
LT-386 Eden



— Nordic Audio
Labs NU-24K



— Sound Devices Astral



It's been a good year for studio microphones, and most of those I've tried have had some outstanding qualities. I was impressed by the scale and ambition of Warm Audio's WA-CX24, the ergonomic innovation of Aim Audio's Inspire, the versatility of Audix's D6X, the honesty of the Royer R12 and AEA N28, and the understated class of the Austrian Audio OC-S10. But the microphone that tested my self-restraint most of all was the **Nordic Audio Labs NU-24K**. It looks stunning and sounds even better, and it will be a long time before my credit card is safe around it!

Sam Inglis

<https://sosm.ag/nu24k>

Wireless radio mic systems are ubiquitous these days, but **Sound Devices** really pushed the technological boundaries with their incredibly flexible **Astral** range, which goes far beyond any other current system's capabilities. For example, 16 digital receivers housed in a 1U half-rack unit is mind-blowing enough, but the ability to tune them anywhere between 169MHz and 1.8GHz, while providing balanced analogue, AES3 and Dante outputs, had me gobsmacked! The associated transmitter units are just as impressively versatile and fabulously well-engineered. The Astral range is not just at the peak of digital wireless technology; it's hovering far above it!

Hugh Robjohns

<https://sosm.ag/astral>

Audio Interfaces & Converters

The **Apogee Symphony Studio** range of USB interfaces reflects the apparently unstoppable trend towards mixing in immersive formats by offering generous output channel numbers and high-end monitor control via a comprehensively specified app. But simultaneously with offering all that, the Symphony Studio range doesn't forget the tracking functions of an interface, so its mic inputs are very highly specified, all but noiseless, and sound really great. Being a USB interface that has both high-end tracking and immersive mixing fully covered makes the Symphony Studio range something of an outlier, and a natural pick for Gear Of The Year.

Phil Ward

<https://sosm.ag/SymphonyStudio>

Having been a big fan of the original Thunderbolt-based PreSonus Quantum, I was sceptical about the reinvention of the range as a series of USB interfaces. Would that not lose what made the originals special? I need not have worried. The **PreSonus Quantum HD8** and its siblings not only offer remarkable audio specs for the money, plus welcome bonus features such as auto-gain and re-amp outputs; they also offer seriously good low-latency performance, at a very competitive price.

Sam Inglis

<https://sosm.ag/QuantumHD8>

RME's ADI range of specialist interface/converters have always delivered impressive class-leading performance, and when the company releases a new model that raises the bar even further it's impossible not to admire their expertise and devotion. The **ADI-2/4Pro SE's** dynamic range performance is the best I've ever measured, and the D-A side improves significantly on its predecessor, too, which is just extraordinary. Digital RIAA equalisation has been added for vinyl enthusiasts, as well as impressive balanced headphone driving capability and a more versatile I/O structure. This is a superbly engineered product.

Hugh Robjohns

<https://sosm.ag/ADI24PRO>

Solid State Logic have been cranking out new products at such a rate this year that it's been hard to keep up. There have been two superb channel strips, plus the excellent 18 USB audio interface and the mighty Oracle console, among others. But the one that has really hit the mark for me is the **Alpha 8**. A line-level ADAT expander might not seem like the most exciting product — but this is so much more than a conventional line-level ADAT expander.

Sam Inglis

<https://sosm.ag/Alpha8>

»



— Apogee Symphony Studio 8x16



— PreSonus Quantum HD8



— RME ADI-2/4 Pro SE



— SSL Alpha 8

» Software

You need a room full of monitors, mounting hardware and cables to mix in Dolby Atmos, right? Well, no, not if you have **APL Virtuoso 2**. Virtuoso 2 is a multi-channel room and monitor modelling app that takes the output from, for example, the Dolby Atmos renderer, and plays it over headphones such that you can be effectively 'fooled' into hearing multiple monitors in a room. Not many *Sound On Sound* review subjects make me laugh out loud but Virtuoso 2 did. It's not dissimilar to the feeling when a good magician performs close magic. Your inability to process what just happened leaves you laughing. I'd listen to an Atmos mix conventionally, then put headphones on with the room and monitors modelled in Virtuoso 2, and the sound would be really pretty similar, and it would make me laugh. Many more times than once I took the headphones off to check the monitors were muted. Virtuoso 2 is truly uncanny. *Phil Ward* <https://sosm.ag/Virtuoso2>

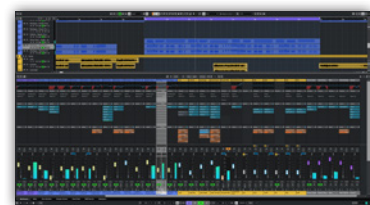


APL Virtuoso 2

I've been a fan of **FabFilter's** plug-ins since they released their Mastering Bundle 15 years ago. Since then, they've rarely put a foot wrong, but this year's **Pro-Q 4** equaliser plug-in is a real highlight. It retains all the good qualities of previous iterations, and adds goodies like resonance suppression and the ability to edit multiple instances from a single plug-in window. I've barely opened another EQ plug-in since I installed it. *Sam Inglis* <https://sosm.ag/ProQ4>



FabFilter Pro-Q 4



Steinberg Cubase 14 Pro



NoiseWorks DynAssist



Strymon Cloudburst Plug-in

A real time-saver, **Noiseworks DynAssist** can make sure every syllable is heard clearly, and every breath and ess is near the optimum level. For something that, when you boil it down to its bare essentials, simply turns your audio up and down, it's an incredible piece of software. It has already changed the way I approach editing and processing dialogue, and a little bird tells me there'll soon be some impressive new strings to its bow! *Matt Houghton* <https://sosm.ag/DynAssist>

I'm lucky enough to have experienced most of the major DAW platforms over the last 20-plus years, and have reviewed

a number of them for *SOS*. There are some amazing options available, but Cubase has been my personal DAW-of-choice for all of that time. Amongst a range of refinements to an already impressive specification, **Cubase 14 Pro** added a brilliant Drum Machine instrument, a cool new Pattern Editor environment and the very powerful Modulator system. My personal 'best' just keeps bettering better. *John Walden* <https://www.steinberg.net/cubase/>

Strymon's Cloudburst pedal is still unique amongst reverb effects, going beyond 'shimmer' to a type of reverb tail resynthesis that makes the reverb sound almost like an orchestral layer. **Cloudburst Plug-in** brings that sound straight to your DAW, making editing easier and of course allowing you to use the effect on multiple tracks if required. Having the reverb enhancement on a continuously variable control rather than two switch positions is really useful, as are the freeze functions. While Cloudburst might seem to be aimed at ambient composers, where it is of course extremely welcome, it can also fit into many other musical genres too. *Paul White* <https://sosm.ag/CloudburstPlugin>

Hardware Instruments

Having already pulled ahead as the most functional production workstation on the market with version 3 of the MPC OS, **Akai** followed up with a mic-drop of a launch in the **MPC Live MkIII**. With four times the processing power, even more expandability, mic pres and a built-in mic, the new Live is finally a true one-stop portable production system.



Akai MPC Live III

It's also been upgraded as an instrument, looking to widen its appeal with step sequencing and clip matrix workflows, and introducing an innovative, expressive pad grid. *Simon Sherbourne* <https://sosm.ag/MPCLiveIII>

What makes a good Eurorack module? Originality? Usefulness? Ease of use? **Cutlass' Gloop** combines all three. The idea of a tape-style looper but with four playback heads instead of one is certainly original, and it's definitely useful... Wondering what to do with your carefully crafted eight-step sequence? Gloop can reorder it on the fly, turn it into a half-speed version, do Steve Reich-style phasing, make it into a textural drone or a lovely twinkly ambient piece — and probably lots of other things I haven't thought of yet. It's also intuitive, blessed with a decent reverb and delay that can affect passing as well as recorded signals, and rejoices in one of the best digital implementations of Friggertronics I've come across. Gloop is very much the gift that keeps on giving, and in my opinion is a very good Eurorack module. *David Gasper* <https://sosm.ag/Gloop>

The follow up to **Melbourne Instruments'** impressive Nina debut, the **Delia** is a six-voice polysynth boasting a 49-note keyboard and of course Melbourne's signature motorised knobs. These don't



Cutlasses Gloop



UNHINGED CREATIVITY.

Seven kits recorded by Sylvia Massy at the iconic La Fabrique studio in France. Expand your Superior Drummer 3 today.





Melbourne Instruments Delia



Roland TR-1000



Sequential Fourm

» just allow the physical panel controls to snap to correct values when loading a preset; they can take on steps or detents and move by themselves to take care of certain types of automation. The Delia is armed with fantastic-sounding DCOs, dedicated high-pass and low-pass filters, onboard effects and a slew of other functions, including the brilliant Morph, which gives each preset two interpolated layers that can be 'faded' between, with its motorised knobs moving in real time.

William Stokes

<https://sosm.ag/Delia>

Ending decades of speculation about whether **Roland** were done with analogue, the OGs are back with a flagship, best-of-both-worlds drum machine that puts the transistor back in Transistor Rhythm. Most of the original 808 and 909 synth circuits are on board, side-by-side with ACB digital, FM, PCM and sampling. Built around the ultra-performable TR-8/8S mixer-style design, but with greatly improved usability, and incorporating modern performance and sequencing workflows influenced by newer-generation instruments, the **TR-1000** is the best drum machine Roland have ever made.

Simon Sherbourne

<https://sosm.ag/TR1000>

At first glance, **Sequential's** latest synthesizer offers nothing new. The **Fourm** is a two-oscillator, four-voice analogue synth with mini-keys. Big deal. But this little powerhouse turns out to be all sweet spot, with a sumptuous vintage sound at a reasonable price. It's a masterclass in miniaturising, somehow managing to retain a big synth analogue sound and user experience. Think of it as a monosynth

with four voices, surprisingly playable polyphonic aftertouch, a one-knob-per-function front panel and plenty of thoughtful modernisations. It also proves that Sequential is in great hands under the stewardship of Focusrite. *Rory Dow*

<https://sosm.ag/SequentialFourm>

Virtual Instruments

Following the rich pickings of new hardware synths and keyboards over the past few years, 2025 has — for an old progger like me — been a bit of a fallow year. But I continue to be impressed by the ongoing advances in soft-synth technology. There are numerous low-cost but high-quality offerings from the likes of Cherry Audio, but the standout for me was **Arturia's** latest V Collection. The company keep adding high-quality synths (innovative as well as imitative) and **V Collection 11** is now so wide-ranging that I doubt that anyone will ever fully plumb its depths. I just wonder when they're going to add all of the other instruments that I've requested.

Gordon Reid

<https://sosm.ag/ArturiaV11>

Since I first used it back in 2023, **Synthesizer V** by **Dreamtonics** has regularly left my mouth contorted between my jaw hitting the floor and a smile so big my face might split. The option to use a virtual instrument that can deliver the melody, lyrics and expression of a very respectable human session singer is a bit of a songwriter's dream. The arrival of **v2** didn't change that one bit, it just refined the software's operation and added some additional functionality. Synth V is a truly remarkable piece of technology. *John Walden*

<https://sosm.ag/DreamtonicsSynthV2>

Beautifully recorded, versatile, easy-to-use samples in the form of loops and one-shots — there's a lot to love about **Bones**, the sample pack containing over 1000 royalty-free sounds by British creative studio **Song Athletics**. When I first heard it, I felt like I'd been sent tracks by a well-curated group of talented performers and recordists with a seriously good ear for tone. You can keep it simple by layering drum loops, melodic lines, and vocal samples, or get properly experimental with modulation effects, distortion, reversing, chopping, time stretching — the studio-quality recordings are a treat to work with. While this is a general-use sample pack containing the basic building blocks needed for a piece of music, there's nothing generic about the sounds and I must say, I've thoroughly enjoyed exploring the collection over the past few months. *Sonal D'Silva*

<https://sosm.ag/Bones>

Spitfire Audio's BBC Radiophonic

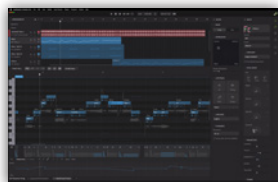
Workshop library celebrates a cherished national institution. Drawing upon a vast tape archive collated by Mark Ayres, this fascinating collection takes a Tardis-like journey through 40 years of musical ingenuity, bringing together experimental 1950s electronica, otherworldly soundscapes, sci-fi effects, early analogue synths, historic instruments, 'found sounds' performed by original RW members at Maida Vale Studios, junk percussion, tape loops and new processed 'warp' treatments. Also included is a brilliant recreation of Delia Derbyshire's iconic, swooping oscillator-driven 'ooh-wee-ooh' lead sound heard in the Doctor Who theme, an essential patch for all keyboardists!

Dave Stewart

<https://sosm.ag/RadiophonicWorkshop>



Arturia V Collection 11



Dreamtonics Synthesizer V Studio 2 Pro



Song Athletics Bone



Spitfire Audio BBC Radiophonic Workshop



Sub51 Drop Pad 3

I was a fan of the original Drop Pad for Kontakt, but it has now been updated to run also on the free Kontakt Player, as well as adopting a number of new performance features. It also has a completely new sound library, though owners of older Drop Pad versions can still use their original samples alongside the new ones to create new patches. Drop Pad isn't the only instrument to allow four sample sources to be dynamically blended, but the ability to add movement also extends to the effects. Its rhythmic patches, which can be built up from layers, always remain in sync and can be locked to any DAW tempo. Despite its flexibility, **Drop Pad 3** remains intuitive in use and is capable of generating sounds that span all genres from ambient to gritty EDM. *Paul White*

<https://sosm.ag/DropPad3>

Guitar Technology

The **Electro-Harmonix Oceans Abyss** can run two types of reverb, along with a range of other effects blocks that can be configured in a number of series and parallel configurations, all aided by a display showing the connections in graphical form. There are delays, filters, modulators and more to play with, making Oceans Abyss something of a multi-effects workstation. It is fairly intuitive to use but digging deep can take time, which is where the upcoming software editor will come in very handy. If you like exploring complex delay and reverb effects, you'll find a lot to like in Oceans Abyss. *Paul White*

<https://sosm.ag/OceansAbyss>

The great thing about a 'Tweed-era' Fender Deluxe amp is that it sounds almost instantly 'right' for a whole lot of the classic electric guitar sound 'vocabulary' the moment you plug into it. Which is good, because there's very little on the amp for you to tweak: the Tone control really only has a couple of sensible settings, and you might want to get fancy by dialling in a little interaction between the channels, but apart from that, you just set the Volume low if you want the beautiful, rich cleans, or turn it up for the uniquely gnarly distortion it offers. **Origin Effects' Deluxe55** pedal version makes it even simpler by omitting the second channel from their incredibly

authentic-sounding, all-analogue recreation of this classic tube amp, which uses solid-state components to replicate every part of the original circuitry. You can run it into an amp, but I think it does its best work paired with a really good IR or speaker modeller, where it can give you the instant gratification of a miked-up Tweed Deluxe in a DI recording. Yes, it will 'flub' like mad under full distortion if you are not careful, just like the real thing, but that's just part of the beauty of this amp — it puts so much range of expression into the hands of the player and the choice of pickup and volume control settings. I've simply never heard a better emulation. *Dave Lockwood*

<https://sosm.ag/Deluxe55>

I often struggle to find musical applications for ring modulators, but the **Red Panda Radius** pedal adds features that really help push the boundaries of what the effect can achieve. With the ability to blend ring modulation with frequency shifting, a flexible modulation section that includes an envelope follower and the ability to have the modulation oscillator track the incoming audio pitch (monophonic), the results go from classic inharmonic ring modulation to bell-like timbres, drones, harmonic enrichment and even slow, dreamy modulation. If, like me, you had written off ring modulation as too unmusical, this pedal might get you to reconsider, and I suspect the modular synth fraternity will really love it. *Paul White*

<https://www.redpandalab.com/products/radius>

Based on the same hardware as the Collider pedal but running different code, **Source Audio's Encounter** is a dual-channel pedal aimed at the production of cinematic and ambient reverbs and delays. The user can combine a delay and reverb or two of each type of effects, where the many options include filtering, modulation, pitch shifting and reverb reverb. I had so much fun reviewing this pedal that Source Audio said they'd like to use some of the presets I'd created. If you like the transformative abilities of granular processing but crave something smoother, then Encounter is most definitely a pedal to try. Interestingly, their on-line editor includes a virtual model of the pedal so you can

create or edit presets even without a pedal connected. *Paul White*

<https://sosm.ag/SourceEncounter>

It's no secret that I'm a big fan of **Universal Audio's** amp emulation pedals — all of them! Even as a diehard lover of classic tube amps, I have to say that I often now use a DI'd UA pedal for recording in preference to the 'real thing'. Sometimes it's just quicker and easier than rigging a mic, or firing up the OX, with no loss of tone or feel that I can discern. Regular readers may recall that UA's Lion 68 'Marshall' was one of my GOTY picks for last year, and it is still my most-used UA pedal, but the **Enigmatic '82** 'Dumble' now runs it a very close second. There is so

much in this 'amp' that you don't find in anything else. It is well known that designer Howard Alexander Dumble would tweak every amp's circuitry to meet the preferences of an individual player, and whilst I certainly can't claim to have the same insight and expertise, I find the comprehensive set of variables in the control app lets you explore every tonal nuance that you

could wish for. The signature 'distorted-yet-somehow-warm-and-articulate' tone is in there, but there's a whole lot more as well. I feel like I've only just scratched the surface. *Dave Lockwood*

<https://sosm.ag/Enigmatic82>

Guitar rig modelling/simulation has come quite a long way since I reviewed the Zoom 503 back in *SOS* October 1998! Some of the top-of-the-line hardware modellers and profilers currently available are truly amazing, but they also tend to come with a top-of-the-line price tag. IK Multimedia challenged that situation with their ultra-compact ToneX One (reviewed by Paul White in *SOS* September 2024) and now **Valeton** have pushed the bang-for-buck ratio even further with the **GP-5**. With a full rig,

and both modelling and profiling options within a mini-pedal footprint, at £75, it shouldn't be able to sound good. But it does. As a compact guitar rig for personal practice, band rehearsals and the occasional gig, Valeton's GP-5 is an absolute steal. *John Walden*

<https://sosm.ag/GP5> **///**



Red Panda Radius



Electro-Harmonix Oceans Abyss



Universal Audio UAFX Enigmatic '82



Origin Effects Deluxe55



Source Audio Encounter



Valeton GP-5

INSIDE TRACK



Zach Nahome & Charlie Holmes ■ Olivia Dean *The Art Of Loving*

To perfect Olivia Dean's intimate, personal sound world, the artist and producer/songwriter Zach Nahome decided to step outside the studio.

PAUL TINGEN

One of the UK's leading producers and songwriters, Zach Nahome's credit list contains prominent names like Bakar, Loyle Carner, PinkPantheress, Slowthai and Celeste. Recently, Nahome hit the big time with Olivia Dean, as co-writer, co-producer and executive producer on the British singer's second album *The Art Of Loving*, and its lead single 'Man I Need', both of which

went to number one. "My process tends to be to go into a room with the artist with nothing, and leave with a song written," says Nahome. "I also tend to lay down the demos. I'm not the best guitarist or pianist or drummer, but I figure it out."

"When I started working with Olivia, she brought in a few songs that had already been written when I got involved, so my job was to help develop and finish them, as well as write new ideas together. Still, the overall process with her was the

same. The record started from within and from an internal dialogue, rather than considering an audience too much. *The Art Of Loving* is an ambitious record, and if your ambition is to really make the best music, the byproduct is that it will appeal to many people."

A Room Of One's Own

"When we were talking about where to start work on the record, she was thinking studios, but nothing felt exciting.



■ Zach Nahome (centre)
with Olivia Dean and bassist
Finn Zeferino-Birchall.

At that point, she found all recording studios quite uninspiring and impersonal. So the idea came about to convert a house into a studio.

"It's quite difficult to find a place in London where you can make noise, but in the end we found a big house in East London that was perfect. We set up a studio in a big kitchen/living area with big massive glass windows looking onto a garden. It felt very open. The house was really cool, and we ended up working there for two and a half months. Olivia even stayed there part of the time. We'd often cook whilst making music, or I'd be in the kitchen making pasta whilst leaving something on loop to listen to. That studio became our biggest tool in

creating these songs. Equipment is great and amazing, but honestly, the room is the most important. By miles."

Nonetheless, equipment remains essential, and Nahome recalls, "We brought in loads. We obviously had an obstacle in that room. The acoustics weren't going to be incredible because there's glass everywhere, so we had some baffles put in and built a vocal booth. I used a pair of Barefoot MicroMain27s, which are really good. We linked two [Universal Audio] Apollo x8ps together in order to get 16 preamp channels. We wanted to be able to just plug in and go. Everything was done in Logic, the DAW I have used for my entire career."

House Music

"We brought Olivia's piano over from her home, and she also brought in her nylon-string guitar, which she loves. Rebecca Ellis at Gibson gave us loads of guitars to play with, which was great, and I also brought some guitars from my studio. We had a vintage Prophet-5 and a Wurlitzer, and two drum kits: a vintage Gretsch and a vintage Ludwig. We also had lots of microphones, and tried several on the drum kit, but I ended up removing half of them because I didn't love the sound of over-miking a drum kit in that environment. It sounded better using less, so I used just a Shure SM57 as overhead, another 57 on the snare, and I think the kick mic was a Sennheiser e902. Electric and bass guitars were recorded DI, which gives me more control over the sounds I want.

"I'm less bothered by what mics there are on the drums, but I care a lot about what mic I use on vocals, and in this case a friend of mine lent me an amazing microphone. All her vocals and all acoustic guitars recorded in the house went through this AKG C12. We also recorded all the backing vocals on the C12, set to omni. We wanted to feel the house. The house was quite noisy, and we wanted that to be audible. The sound of the house became part of the sound world for the album, one of the common threads that holds it together.

That C12 also was a really big part of this record. I ended up buying it, because of how important it became to my process. That particular mic has a lot of history. It used to belong to Olympic Studios in London and then to Dusty Springfield, who recorded most of her songs with it. It's incredible.

"We also often used the Teenage Engineering TP-7. It's almost like a tape recorder. There were a lot of noises around the house in East London. It was next to a school and the garden had amazing birds. I recorded a lot of the sounds of the environment that we were in with the TP-7. I also use my iPhone all the time. Sometimes I record vocals or drums solely through Voice Notes. The amount of times I'll use the iPhone for recording guitars and pianos is amazing. Also on this album. Many of the songs may sound lush and big, but there's this weird intimacy that you get from how your iPhone makes things sound, that spoke to the personal element that we were after."

Less Is More

Many other writers and producers are credited on *The Art Of Loving*, including Bastian Langebaek, Max Wolfgang, Matt Hales and Tommy Danvers, and American production and writing heavyweights like John Ryan, Julian Bunetta, Amy Allen, Tobias Jesso Jr and Nolan Lambroza. According to Nahome, most of their work on the songs took place before the East London sessions.

"As I mentioned, some of the songs pre-dated her and I meeting. For example, the year before, Olivia had worked in America on a song with John Ryan and Amy Allen. Julian Bunetta and John later also offered some work on the production of songs like 'Close Up' and 'Let Alone The One You Love' on the drums and brass, in particular, which was great. Moreover, Olivia has worked with guys like Bastian, Max and Matt for a long time, and they are an integral part of her writing process. They make great songs together.

"Some of these songs, because of the idea of the record being super-personal, had to be on the record. Tobias Jesso Jr, »

Zach Nahome: "I think building an album is much like making a movie. There are all these threads and subplots, and even if they're subtle, they're important."



■ Much of *The Art Of Loving* was written and recorded in a London house rented especially for the purpose.

» Max Wolfgang and Bastian also worked at the house in East London, and some of the additional production from John and Julien took place around that time remotely. All songs went through my computer at some point or other, and we tried to elevate them. For example, when I first heard 'Let Alone The One You Love', I said to her, 'This feels like it wants to be in 6/8.' That gave it the sentiment that I understood she wanted, so we changed the rhythm.

"In general, I often didn't redo or replay or change things. I did add stuff, but also often took away stuff. For me, good production often is reduction. I'm constantly trying to remove and declutter stuff. On a record that's so personal and where honesty and vulnerability are at the core of it all, removing stuff really allows that to shine through."

Strong Feelings

A number of songs were also written from scratch in their makeshift studio. "Man I Need" is a great example of something that was written from scratch in that house, with Tobias Jesso Jr, who came over from the US for a week. We were having a conversation around relationships for half a day, and that

turned into a song, effortlessly. It started with Olivia playing some chords on the piano, and I went to the drum kit set up in the middle of the lounge. We mapped out the entire song, and at a later date Olivia and I sat down and tried to find chords that felt better. The pre-chorus chords on the original demo were the same, but the chords throughout the rest of the song changed.

"The demo sound is not that dissimilar to the final sound. Obviously, it got added to and built on, but it was broadly the same world, with the same feeling. I did the sonic palette and the choices of guitars and keyboards, and the way everything's layered up. So there's Olivia's piano, and I played the Prophet-5, and the Wurlitzer, as well as guitar and bass. The drums are a combination of live drums and machine drums. I would have usedAlchemy or Ultrabeat, I can't remember. I love the Logic stock plug-ins and instruments. I use them all the time.

"About half of Olivia's vocals were recorded at the house, with me doing vocal production. I'm massively obsessed with that. Olivia and I did a lot of takes. Generally, she likes to record her vocal parts from the beginning to the end of songs. She'll do that as

often as she feels she needs to, to get what she wants. And after that we'll sit side by side in front of the computer, going through and comping.

"Olivia is incredibly hands-on in the studio, which is one reason why I love working with her so much. There's no more harsh critic of her vocal takes than Olivia. If she's not happy, she'll do it again. If we need 100 takes, we'll do 100 takes. There's no tuning. We tune through editing. That's why we're so serious about the vocal comp. In terms of the vocal chain, I try to keep it as neutral as I can. I'll have a compressor, an EQ and a reverb, but I don't have a set vocal chain. For every song I'll build the chain from scratch because I don't want to be complacent and think that a particular channel strip suits every vocal and every song.

"On some songs Olivia worked with Max Wolfgang, who vocal-produced as well, because it was just more time efficient. Max has an incredible understanding of vocal harmonies, and helped a lot with that, also with the backing vocals. They were recorded by Olivia and her band. She's known these guys for 10 years, and went to school with many of them. They're really close,

and there's a big chemistry between them. It felt right to have the humanness of a band who are incredible players, but perhaps not the best singers, doing backing vocals with one mic in a room. They were all standing around with headphones on, and a lot of the time they were laughing and being playful."

Plot Lines

Additional recording sessions took place at RAK Studios in North London, including all string and brass recordings and some band sessions. "I was hellbent on getting in a string arranger right from the beginning of starting work on the record. We asked Rosie Danvers to do them, who is incredible. Olivia and I knew what we wanted, so we hummed the parts we had in mind in front of Rosie, and also the horn players, and explained what we wanted. Rosie also interpreted and performed parts in her own way and brought her own ideas, which were great.

"For me these string and brass parts were another aspect of building a record that has subplots and threads that run throughout, like the sound of the house in East London. I think building an album is much like making a movie. There are all these threads and subplots, and even if they're subtle, they're important. It's what creates cohesion. Another thing that unifies the album is that it's a short record with short songs. Especially on an album like this, where the sentiments expressed meant that it needed to be as concise and uncompromised as possible, we didn't have space for filler or fluff. My principles in making both songs and albums are the same. It's always the question, what can we delete? In removing stuff, you're left with what you need the most. And you can make that shine more."

The Final Stretch

Nahome was rough-mixing the material throughout, and towards the end of the process, he took the sessions back to his own, minimalist studio to create final rough mixes to send to mixer Charlie Holmes. "My studio has to be really well treated, because I love big low end. I have it in my DNA, because when I was a kid, my dad was obsessed with sound systems, and in his car he had the loudest available subwoofer. And true enough, my first car, a small Fiat, had the biggest subwoofer that I could fit! I've had subwoofers in my own studios

from the beginning. My monitoring today is hilarious, because I have a set of Neumann KH 120s and a set of KRK V6s, but my main monitors are the ATC SCM20As, with the KRK 10S sub.

"The rest of my gear is simple. I use the Prism Atlas as an interface and for preamps. It's the best. I have a few Neve preamps, and some great mics, but I don't have or use a lot of outboard. I just want the best converters and preamps. The Prism gives me that. For the rest it's about great mics and great recording technique. I do have many instruments in my studio, including loads of guitars. I have a piano. And still I have my first drum kit from when I was 11, which is a Pearl Export.

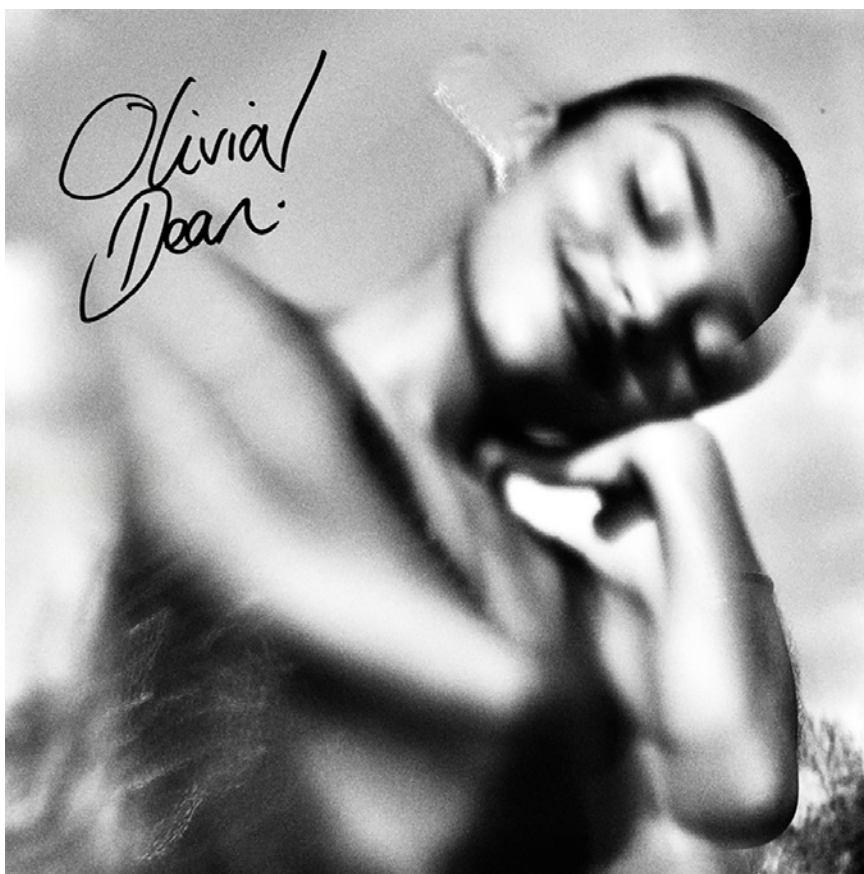
"Low end was also a huge conversation with Olivia's album. We talked about this at the beginning of the record, and she said that she wanted this record to sound really fat and thick. The joke between Olivia and I throughout making this record was that everything sounded too fat. We'd always laugh at that. She'd look at me and be like, 'Come on, surely not? This is ridiculous!' Every time there was a mix, before Olivia would even listen to it, she'd know that the bass and the

kick would need to come down a bit. But that's a better place to be than the other way round.

"With the final rough mixes for Olivia's album, she went to America for a week, which gave me time to really get into them, and also to add some additional production. I added a number of bells and whistles and got the mixes really nice. When she returned, we sat together and went through them all and did some more tweaks and anything else that needed to be done. Once we were both happy, we sent them to Charlie. I send him stems, because he doesn't like Logic. He's a Mr Pro Tools.

"Charlie also mixes with his ears and is also obsessed with low end. He doesn't overcook things, and has good taste in not putting on too many effects. With this project I did not even send him notes. I just let Charlie figure it out for himself. He understood what the record needed and there were very few recalls because he got it right so quickly. We've done quite a few projects together, and I was adamant that he mix this record. Olivia trusted me on this and was as blown away and happy with the mixes as I was." ■■■

»



» **Charlie Holmes: Mixing *The Art Of Loving***



■ Charlie Holmes in his minimalist mixing space.

Having been mentored by Mark ‘Spike’ Stent, mix engineer Charlie Holmes now works from his own room near Salisbury. “I have a very simple mix room, with Neumann KH 150 monitors and a sub, Pro Tools with tons of plug-ins, Dangerous Music Monitor ST controller, and SSL UC1 DAW controller. My room is small and has loads of acoustic treatment and is pretty flat. For whatever reason, I find it easier to get the bottom end to sound tight and punchy in a small room.”

Holmes asserts that he’s “quite an untechnical mixer. A lot of what I do is about feel. When working on Olivia’s record, she said, ‘I want it to sound like a warm hug all the time.’ If you do tons of processing, you won’t get there. So I was very minimal in my use of plug-ins, and just did small tweaks everywhere. And we did not get into crazy revision numbers. I tend to mix top-down, so I start with everything in, get an overall balance, then focus on the groups, and then home in on individual tracks. If you start your mix by soloing stuff and getting into tiny EQ moves too early, you can get bogged down and lose the big picture.”

Holmes dives into ‘Man I Need’ as an example of his approach. “Because

the production on ‘Man I Need’ was a little more minimal than the rest of the album, it needed less heavy lifting. It also was one of the last songs I mixed for this project. I remember being absolutely shattered, and coming into my mix room feeling dreadful. But when I pressed play and heard that song, it was unreal. I was jumping round the studio, which is not like me at all. It felt great straight away.

“In mixing ‘Man I Need’, my main challenge was, ‘How can I get as much of that energy across as possible?’ There was quite bit of the sound of the house on the live drum stem, and on the vocal stems. I could hear background noises, and on the drums I was trying to preserve them with compression — with vocals you don’t want to bring it up too much. Finding the right balance was a challenge in this mix.

“I heavily compressed the live drum stem to bring out as much energy as possible, using the UAD Chandler Zener Limiter. I find that you can really crush live drums with this plug-in. It adds a lot of character. Similar with the UAD Neve 33609 on the drum parallel, that brings out more transients. The track had machine drums and live

drums that both went to the drum bus, on which I did not do crazy amounts. There’s the Waves PIE compressor, which I tend to use on my drum bus as it brings a live kit together like it’s one instrument rather than loads of mics.

“On this track I wanted the sample drums to feel more natural and more together with the live drum room track, to tie them in a little more with the vibe of the album. The PIE also allows the percussion to pump subtly to enhance the groove. The Overstayer adds saturation and does something lovely to the bottom end, and to the snare. It also adds some nice weight. hilariously, I’m then cutting some low sub with the Dangerous Bax EQ, probably as a result of feedback I got that the track was getting a bit too heavy. The song has to be all about Olivia’s vocals!

“When working on lead vocals, I tend to work with the dry vocal and two parallel tracks. This is especially the case with an artist like Olivia, because she doesn’t like her vocals to sound too compressed. She wants her vocals to sound natural, but you still want to it to have loads of energy and cut through. It’s a radio record, right? So there’s the main

vocal, with just 2-3 dB compression from the UAD LA-2A, but then there’s a parallel with 20dB gain reduction from a UAD 1176, with the fastest release and a pretty slow attack. If I want more urgency and for the vocal to be more up front, for example in the choruses, I bring up the parallel, and I pull it back again in the verses.

“There’s another lead vocal parallel, with the Soundtoys Devil-Loc, applying some more compression, but the Darkness button is dialled down, so it’s not adding more ‘s’ and ‘t’ sounds. I mixed this in very low, volume-wise, but again, I pushed it up in some sections, so I could adjust the character of the vocals for these sections. These three tracks feed the lead vocal bus, on which I had the SSL SL4000E, which I use for everything. I think it’s really good to have a plug-in that you know inside out, so you know exactly what it’s going to do. I used to have the UAD version, but when SSL came out with their controller, I started using that.

“The other big thing on the vocals is the UAD Lexicon 224. There’s an ‘80s vibe about this song, but it also had to sound modern. So I used a long gated vocal verb, and some



Sampled and real drums were sent to a stereo bus to be processed by this signal chain.

Soundtoys MicroShift, and then a lot of tape slap from the EchoBoy. On the All Lead bus there's some Oeksound Soothe and another SL4000, and some Slate VMR, adding air. I hate harsh 3-4 kHz frequencies in vocals, so love Soothe. All plug-ins are doing light touches. It may be difficult to hear what difference each makes, but if you turn them all off, it's a big change. Some of the backing vocals have the Arturia Rev Plate-140, doing very little, and a FabFilter Pro-Q 4 before it for some subtractive EQ. I use the Q 4 only

for surgical subtractive stuff, and the SL4000 for boosting, because of the character it adds. It's punchy and adds some harmonics, which is really cool.

"I very rarely use much compression on the mix bus. In this song I had The Oven, the SSL Bus Compressor 2, and the Virtual Tape Machines from Slate. The Oven adds a tiny bit of saturation in the lower mids and a bit of air as well. I had a mix note at one point that the drums were too heavy, so I added the SSL Bus Compressor because it kept the

energy of the drums, but stopped them being quite so aggressive. It's doing 1.5-2 dB, absolutely maximum. The Virtual Tape adds some of that midrange warmth that Olivia wanted for the record. That particular tape plug-in does a thing where it pulls the lower mids forwards a little bit, which I quite like on the vocals and the snare. It just adds a bit of weight to things, in a nice way. It makes them punchy rather than muddy.

"Finally, there's the Schwabe Digital Gold Clip. I love that plug-in.

As I said, I don't love heavy compression on the mix bus, and I don't love heavy limiting. But Gold Clip is fantastic, especially with transient-heavy stuff. You can get volume without having to smash transients. It's clipping them, but you don't get pumping. It keeps the feel the same, and at the same time gives a lot more volume. At the end of the day, that's where we're at today. Again, there's nothing crazy going on. They're not big moves. It's the energy and feel of it."



The UAD Neve 33609 compressor plug-in was used in parallel on the drums to add some transient attack.

TalkBack

Donato Di Trapani

WILLIAM STOKES

Sicilian composer, producer and instrumentalist Donato Di Trapani is best known internationally as synthesist and keyboard player for Paolo Nutini, but he is also the co-founder and resident producer at Indigo Studios in Palermo, where he and Nutini first met. The composer is currently working on a suite of electronic tracks reinterpreting vintage recordings of Sicilian folk songs.

At the moment I can't stop listening to

There isn't really a record that I can't stop listening to right now. There is, however, a record that I've never stopped listening to, and that is Bach's *Goldberg Variations* played by Glenn Gould. The second recording. I almost know it all by heart. I studied classical piano between the ages of seven and 15, but as a teenager, I wanted to create, not just execute. However, what I retained from that experience was Bach. In terms of melody, everything I've learned I've taken in some way from Bach. I believe that the *Goldberg Variations* represents one of the highest points of Western culture in terms of building melodies and their variations. Glenn Gould's interpretation was so amazing, using almost no sustain pedal, as if playing a harpsichord. It's so clear. He made two famous recordings, one in '55 and one in '81. The 1955 one is incredibly fast and virtuosic; it's so impressive in terms of technique. The 1981 one is much slower, but it's almost summing up everything he had learned about music. He died around one year after that recording. It's so moving. When I'm composing in the studio, Bach is always one of my biggest influences.

The artist I'd most like to collaborate with

If I could choose, I would do a soundtrack for a Werner Herzog movie. I like the way he approaches his art in this



uncompromising yet still magical way. And in every movie he does, there's such a strong kind of intuition in the ideas. For example, *Nosferatu* — it's a Popol Vuh soundtrack — starts with a crazy choir of very low voices, and then it completely changes into this very peaceful guitar track. I find it very powerful. I would like to collaborate with someone who has that kind of sparkle, of real magic. In terms of musicians, there are of course tons I would like to collaborate with! One of them is undoubtedly Nicolas Jaar. I love the way he went from kind of a 'superstar DJ' to someone who turned his back on the system and started approaching things in a very personal way, focusing on his own cultural and musical roots, involving local communities all around the world. His work is simultaneously poetic and political.

The first thing I look for in a studio

I mostly get excited when I see gear that I don't know; something I've never seen

in my life, that I don't know how it works. It forces me to come up with a more creative and instinctive approach to what I want to do. On the contrary, if I go to a studio where I find all the classics, like a [Sequential] Prophet 6, a Minimoog, a [Roland] Space Echo... I'll be like: "OK, I know how this sounds. I know how it works." I'll have to be really inspired by the music, by the project, by other things. It won't be the gear that's inspiring me. But if I find something that I've never used before, and maybe that I have to struggle with a bit, that can be inspiring: a prototype of a synth, for example. A while ago I went to the Museo Del Synth Marchigiano, an amazing place in Italy where a lot of the instruments are kind of a prototype. So you essentially have to guess how it works! That's really exciting for me.

The person I would consider my mentor

It's a Sicilian artist who was and still is very famous in Italy: Franco Battiato. »



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Grammy-nominated mixer and producer (Venom: The Last Dance, Alien: Romulus, God of War, CeeLo Green)



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» He died in 2021 and was a unique figure. He was a hybrid between a musician and a philosopher and had such a special take on music and life in general. I got the chance to have lunch with him back in 2011; I was telling him about my own experiments with my first band, and he was giving me some philosophical advice, like "Always follow your will." It was of course very generic but at the time it had a strong impact on me. Only later did I realise how right he was! In every interview, he had this kind of detached vibe and humour, which is very Sicilian.

He also did many avant-garde electronic experiments on his first records, which are amazing. I was listening just last week to an old album, *L'Egitto Prima Delle Sabbie* [*Egypt Before The Sands*], composed in the late '70s: it's just piano loops, evolving for like 20 minutes. But then, all of a sudden, he said, "OK, now I want to become famous, I want to be known to a lot of people. I'm going to make a pop record." And he made what is, for me, the most amazing Italian pop record ever, called *La Voce Del Padrone* [*The Master's Voice*]. It's a short record, seven songs, but very dense, with genius songwriting and very smart synth arrangements. It's one of my favourite records ever.

My go-to reference track or album

The reference for what I am composing right now has definitely been a compilation called *La Ola Interior: Spanish Ambient & Acid Exoticism 1983-1990*, released by a Swiss label called Bongo Joe Records. That was an incredible discovery for me. I didn't know that in the late '70s and early '80s, there were so many artists experimenting with electronic music, but at the same time sounding so Spanish in the way they were sculpting things. For example, the drum sounds aren't just 'drum sounds' most of the time, they are sequenced synth percussion with a very warm low end and a peculiar tone. All the synth lines, they are not the 'classic' synth sounds you hear from the '80s like the Human League or Depeche Mode, or whatever. It's a different harmonic texture.

My secret weapon in the studio is

It's an ability, really: I think I have a very quick and instinctive way of associating sounds with what they can be used for in a piece of music. And it's interesting,

because it can make my work move much quicker. I can listen to dozens of presets, and sometimes I'll hear one sound and immediately say, "OK, this is the sound." This could be a sound for a riff, or a percussive part, but it's something that I've always had. And I think that's been kind of a secret weapon, because it allows me to interact with a project, or with people, in a very instinctive way, without having to think too much.

The studio session I wish I'd witnessed

OK, I'm going to go pretty 'pop' here, but I have to be sincere: I would have loved to witness *Kid A* being recorded. For me, it's one of the biggest mysteries of contemporary music: how this incredible rock band went from one of the most amazing rock records of all time to an electronic record in which there are so many hints of how things were going to sound in the future. Vocoder, drum machines, all these amazing synthesizers... I always wonder, how did that all happen in one record!? It all changed dramatically, but in a direction that was so influential to everyone. It's a big mystery to me. The last track, 'Motion Picture Soundtrack', has always been a massive reference for me. You have this very sad music, and all of a sudden there's this explosion of, like, little twinkling stars! Listening to that at the age of 15, I was just blown away. The fact that it's possible to mix that level of writing with that kind of arrangement still inspires me. Everything on that record is just a great idea.

The producer I'd most like to work with

I would just love to work with anyone who can really open up new territories for me. Maybe Tony Visconti, while he was recording *Electric Warrior* with T Rex. How did he get to that sound? And in 1971!? So clean, so definite, and at the same time, so powerful. But in terms of producers working nowadays, I love AG Cook, particularly what he was doing with the label PC Music. That way of creating unheard sounds that just didn't exist before, and that sound so contemporary. I like the idea that pop can still be avant-garde too and set the bar for new standards. The producer that would be closer to what I'm doing would be, of course, El Guincho. What he accomplished with his records and Rosalía and the contamination between Spanish tradition and electronic was a huge inspiration to me.

The studio experience that taught me the most

My first session with Paolo Nutini in Brussels, recording *Last Night In The Bittersweet*. It was such an adventure, in a high-level environment, and it was a massive test for me as a musician. I got the message asking me to go to Brussels and once I got there we started recording straight away and didn't end until about seven in the morning. I played so many songs, and they were all so different. There were ballads, Krautrock-like songs, rock songs, there was experimentation, almost avant-garde stuff... everything. So I had to give it my all, relentlessly. And it was funny, there was no comfort zone, no MIDI quantisation, it was all live. There was no real time to think much about ideas for the record, for the songs, because you had to play straight away. There was the producer Dani [*Castelar*] going: "OK, let's do it!" and you had to play. I had so many synthesizers there, I decided on this weird hybrid approach with a Memorymoog, a Juno-60, and a Philicorda, and then I had my usual gear as well, like the [*Elektron*] Digitone and the Prophet 08. Of course, there was a Hammond and piano and everything. But I had to be so focused on every song, to figure out which instrument would work best, to think about the structure and the chords with no charts. It was really another level for me in terms of pressure, of proving myself in an ultra-professional environment.

The advice I'd give myself 10 years ago

Find your sound. It's not about buying expensive gear or, I don't know, even collaborating with cool people. It's about your approach to sound and to music. That's massively important to me: an amazing session pianist can be replaced by any one of hundreds of other pianists. That was a bit of the thing for me with Paolo, at the beginning. I would find myself thinking: "Why is he choosing me? Among all the many talented and amazing musicians he could choose from, and who would probably even play for free." But as time passed and we kept collaborating, I realised that maybe he just liked my approach to sound and music, and that it really matches what he does. Once you have that, you can't easily be replaced. The thing I love the most is when people tell me, "Ah, this really sounds like you," because it means my sound represents me, so that people can recognise me through my work. So that's what I would say: focus on finding your sound. ■■■

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Classic TRACKS

William Basinski *The Disintegration Loops*

William Basinski's archival recording project gained a life of its own when the tapes began to deteriorate, and then took on a new significance in the wake of the September 11th attacks.

TOM DOYLE

One of the most celebrated tracks in ambient music, William Basinski's 'd1p 1.1', the epic, slow-moving opener of his four-volume 2002–3 album series, *The Disintegration Loops*, is a 63-minute-long recording of a 12-inch piece of tape sonically degrading in real time.

Back in July 2001, when it was created, 'd1p 1.1' was in fact the result of an accident. That summer, Basinski was in his Brooklyn loft attempting to digitise some of his old quarter-inch tape loops he'd made in the '80s when he noticed the deteriorating effect and kept his CD burner running. The recording later took on a greater resonance when it soundtracked Basinski's single-take video footage, shot from his roof, of the Manhattan skyline in the wake of the September 11th attack on the World Trade Center.

The combination of the visual and sonic elements — smoke billowing across the damaged city, as slowed brass notes from a random snippet of radio-broadcast muzak gradually decayed — became an unexpectedly powerful one. What had started life as an artful experiment became a poignant audio-visual tribute to 9/11.

"Yeah, this is where it was like, 'Oh God, the music has changed,'" Basinski tells *SOS* today of the moment he realised that he'd created a monument to a tragedy. "The whole world had changed. It was like, 'This is... well, it's an elegy now.'"

In its video installation form, 'd1p 1.1' is now a permanent exhibit at the National September 11 Memorial & Museum opened in 2014 on the original site of the World Trade Center. Basinski admits that he hasn't visited the museum, believing he would find the experience too emotionally overwhelming.

Photo: Sean Stout

"I'm never gonna go see that," he stresses. "I can't deal with that."

Nonetheless, it's further proof that 'dlp 1.1' has only grown in stature in the two-and-a-half decades since its creation. Now Basinski has overseen a new, four-vinyl box-set remaster of *The Disintegration Loops* that reveals all of its strange and fascinating details anew.

"Well, y'know, people discover it when they're ready for it," Basinski says of the enduring appeal of *The Disintegration Loops*, before adding with a laugh, "and then they can't get enough."

Beginnings

William Basinski, born in 1958 in Houston, Texas, took a circuitous route into ambient music. His father was a scientist for NASA, working on the lunar module, and so he grew up in Clear Lake City, a planned community populated by many space program workers, including astronauts. When his dad was relocated to Florida, the family went with him.

"I spent a lot of time practising my clarinet because I was quite a sissy," Basinski says, "and it kept me somewhat out of trouble from the bullies in Florida. Then in '72 when they cancelled the moon program, we ended up moving to Dallas.

"My parents put us in this very fine school district with a super fun, rigorous music program. In high school I played in the jazz band. I changed from clarinet to tenor saxophone."



Photo: James Elaine

Basinski went on to study saxophone at the University of North Texas in Denton. "I totally screwed my auditions, so I switched to composition," he says. There, one contemporary music teacher opened up the future avant-garde composer's mind. "This was my favourite class," he remembers, "because he taught us how to listen. He

took us out into the fields and made us stretch our ears with deep listening. Then we learned about Cage and all the other possibilities that were available with radio, silence, noise, prepared piano."

At the same time, Basinski's fellow students were introducing him to the work of other experimental composers

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» who were to prove inspirational, such as Steve Reich, Terry Riley and Philip Glass. “I got to hear everything that was new and cool and far out. Y’know, Steve Reich’s tape loop studies. And then when *Music For 18 Musicians* came out [in 1978], where he transitioned that merging and phase-shifting into the ensemble, that was just so beautiful, I thought.”

It was at university that Basinski first conducted his own tape experiments, ingeniously sellotaping over the erase head of a cassette-recording Sony Walkman so that he could create long, improvised and live-overdubbed electric piano compositions.

“The first piece was probably around eight to 15 minutes,” he recalls. “We always had a piano in our house growing up, so I could play the piano to compose. I was sort of working on this thing for some time, for some years. I would just, y’know, smoke some weed and fuck around and then try things, and just randomly roll back and intuitively do this other part and see what happened.”

San Francisco

It was through these initial tape experiments that Basinski first met his long-term partner, visual artist James Elaine. “I was in this very snooty art group of kids the second year after I moved away from the dorm,” Basinski remembers. “Jamie was like the king of this group. He wanted to hear about my music, and so I played it for him, and he said, ‘You’re a genius.’”

“So that’s so attractive to hear when you’re 19 years old,” he chuckles. “Very encouraging. So, he fell in love, and I fell in love, and I ended up leaving and moving with him to San Francisco on Halloween, ‘78.” The pair ended up living in a six-roomed apartment on Haight Street where Elaine proceeded to fill every spare room with art installations.

“He would collect these old ‘50s TVs off the street and bring them home,” Basinski explains, “and we’d turn them on and see what they would do.” Rather than their visual content, he was more interested in capturing the sounds these decrepit TVs produced. At the same time he’d become a collector of old reel-to-reel tape machines.

“So I started just recording everything,” he says. “I got some junk tape decks, these big old Philips or Norelco Continental 40lb portables. You could slow the Norelco way down — seven-and-a-half IPS, three and three-quarters and then one and seven-eighths. I just started experimenting

Photo: Sean Stout



Basinski used D-cell batteries to tension the loops on his Revox G36 tape recorder.

with changing speeds and seeing what kind of frequencies I could pull out of these drones I recorded, from the TVs to the refrigerator to everything.

“The sounds in San Francisco were just such a rich ambient environment with the clickety electric buses and then the cable cars creaking, and then you’d have the fog horns. Just really, really rich.”

Another highly influential album for Basinski was Brian Eno’s proto-ambient classic *Music For Airports*, released in 1978. “It was sort of like... the melancholy, the delicateness of it, so beautiful. And I was like, ‘Wow, this is amazing.’ And then hearing Fripp and Eno, and then hearing all this drone music from Germany...”

Basinski subsequently built his own version of Robert Fripp and Brian Eno’s Frippertronics tape-delay looping system. “I started getting some newer quarter-inch tape decks. I had the two flatbeds to loop with, mixing into one of the Sonys or something. I did a lot of that, and then short-wave radio experiments with loops.

“I wanted a Mellotron, so I was trying to make one with these little string snippets. Slowing them down and stuff, you would find this huge well of melancholy. And it was like, Oh, OK, here we go.”

New York

Basinski and Elaine then moved once again, in 1980, to New York. “Jamie was a very talented painter, and he’d had some shows,” says Basinski. “We’d both actually done some shows together, in some very cool galleries in San Francisco, but it was a very provincial scene. We worked and

saved as much as we could, and then moved to New York because he wanted to ‘make it,’ y’know.

“Starting first day there, there was a transit strike, April Fool’s Day, 1980. We had to walk all around town looking for a loft. We took off our brand-new cowboy boots that night and got some band aids and tennis shoes for the next two weeks to walk around [laughs].

“We did eventually find this big, 5000-square-foot loft on the sixth floor in downtown Brooklyn, right across from a big subway station, and near to Brooklyn Heights. It was fairly safe, not too scary, if you didn’t go the wrong way. My friend Roger [Justice] eventually came and moved in with us, because we needed three people to afford the rent. They would be painting, and I would be painting with sound and experimenting with my tape loops and delay systems.”

During Basinski’s early years in New York, he produced the music that was to feature on his debut album, *Shortwavemusic*, made in 1981 and 1982, but not in fact released until 1998.

“Nobody wanted it,” he says in regard to the zero record label interest he attracted at the time. “They weren’t buying what I was doing. It was all punk rock and no wave. We dug all that shit. We went to CBGB, we went to shows and stuff, and I played in a lot of bands too, keyboards and sax and stuff. I just kept doing my work and then played with other people. It took forever, because no-one was interested.”

The short-wave radio sounds that Basinski captured on the album came

from various remote sources. “I made a huge antenna on the fire escape, so that you could really get the signals from all over the world. Just taking the wire around and around and around. You could hear Russia and the Middle East and everywhere. In-between stations was what I liked the best, the particle showers and all that stuff. I became fascinated with those sounds and the way the station might shift on you when you were recording. I just had a field day with that for a year or so.”

Aside from a recording studio and art space, between 1989 and 1997, Basinski and Elaine also used their loft as a club venue they named *Arcadia*, where many of their musician friends, including Anohni and Diamanda Galás, would regularly perform. “It was really something,” says Basinski. “It only really lasted for a few years because it was exhausting. It was in our home, and eventually it started to get to be too much.”

“Diamanda did a fabulous Halloween show [in 1991], which I have an amazing recording of. I had to rent her a Steinway, and so we lost money on that show. But y’know, you can’t have a party like that for \$600, I’ll tell you that. There were already a lot of artists and people that moved into Williamsburg at the same time we did. So it was sort of like a company. We featured the people that we liked over and over again.”

Disintegration

Nonetheless, by 2001, William Basinski found himself broke and on the verge of giving up on his music career. “I was in debt and about to be evicted,” he remembers. “On this beautiful day in July, I was miserable about what I was going to do.” Basinski was sat around idly reading British-American philosophical author Alan Watt’s 1957 book *The Way Of Zen* when he suddenly felt inspired. “I read it,” he says, “and thought, OK, use this time to do your work. Get back in the studio.”

“By then, I had bought a CD burner on the credit card, and so I decided to get these loops transferred to digital. Because, y’know, the tapes were from the ’70s, and the loops were from the ’80s. I had recording engineer friends. I knew what dropout was, and I knew what could happen.”

At the time, Basinski’s old tape loops had almost become decorative,

having been left hanging for some years on a dead tree branch he’d found in a parking lot. “I would group the ones I wanted together with little safety pins on another side.” He picked off six of these loops, each roughly between 12 and 16 inches in length, suspended them from a boom stand, and then randomly selected one.

“That was what became ‘d1p 1.1,’” he says. “I was like, Oh my God. This is so grave. This is such an amazing melody.” All of the tapes contained chance snatches of audio from a CBS radio station broadcasting from the top of the Empire State Building, slowed to three and three quarter IPS. At this point, Basinski’s main tape recorder was a Revox G36 that he’d picked up in a nearby house clearance in Brooklyn in the late ’90s.

“It was this townhouse in the Hasidic section of Brooklyn about a mile from my house. And the man was a very interesting guy. He had worked for Bell Telephone since, like, the ’30s. He had all of these old headsets and microphones. I got a portable cutting lathe and this Revox, which looked kind of ratty.”

With the Revox, Basinski’s technique involved using D-sized batteries to spool the tapes around and maintain tension. “They’re heavy,” he says, “and you’ve gotta keep the tapes tight, or they’ll get chewed up in the machine.”

Listening to the woozy sample repeating in the first loop, he decided to create a counter-melody to it using the in-built arpeggiator in his Octave Plateau Voyetra 8 synthesizer, leaving it to play live and unsynchronised along with the main theme.

“I got a cool, randomly arpeggiating sort of French horn sound dialled in,” he recalls, “and it was going really nice. I didn’t know what was going to happen, so I started recording. And then, I went to make a coffee, and came back and started realising, something’s changing, what’s going on? And then I started to see what was going on and hear what was going on.”

Basinski looked at the moving tape and could see that it was shedding the iron-oxide coating from its plastic backing. “I thought, Check the levels... check the levels. Gotta make sure, y’know. Let’s just see what happens, because this is a one-time thing.”

Adding to the haunting effect was the outboard Lexicon reverb that Basinski

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» had used to treat the sounds. As the notes cracked and slowly disappeared, the Lexicon eerily filled in the gaps. “You can hear it more because the dropouts seemed to happen at the end, where the decay on the notes was. The strongest notes seem to hold on the longest. Kind of like people’s memories and in dementia and things like that. So fascinating.”

After recording the first loop for just over an hour, Basinski moved on to the second one, creating ‘d1p 2.1’ and so on. Over two days he produced the six pieces that were to make up the four albums of *The Disintegration Loops*.

For ‘d1p 2.1’, Basinski had similarly added a sequence on the Voyetra 8, although he chose not to add any overdubs on the later loops. “The second one started to flake out faster,” he says, “and I realised, Listen, don’t do counter melodies here.”

“Some of them were so perfect and eternal that I knew they didn’t need anything. After that, it was just the loops by themselves. I realised, Stay out of the way. This is not your business. You just do your work as an engineer here.”

Over the Summer and into the Autumn of 2001, Basinski and his friends visiting the loft often listened to these disintegration recordings, purely for pleasure. Initially he had his doubts about releasing them. “I didn’t really know if I could just put this on for an hour and call it my work,” he says. “But we would listen to them.”

“Anohni came over. Steve Roden was in town. My wonderful artist friend Howard Schwartzberg, with his amazing Coney Island accent. We got stoned, flopped around the loft and listened to the whole thing, and he’s like, ‘Billy, you’ve done it.’ He was right.”

But then came the day in history that would be forever linked to ‘d1p 1.1’ and *The Disintegration Loops*. On the morning of September 11th, 2001, William Basinski woke up planning to travel into Manhattan for a job interview as administrator at an arts organisation whose offices were located on one of the upper floors of the World Trade Center.

“Thank God it was at 11 o’clock in the morning,” he says. “The towers were gone by then. I mean, it was horrible. We saw everything from the roof and from my window, and it was just... I mean, you just... you never get over that.” One of Basinski’s friends had set up her camcorder on the roof, video’ing the city and clouds of smoke.



■ The tape loops after the recordings.

He asked her if he could use his own tape to record the last hour of daylight before night fell. “I had her help me frame it up,” he says. “And I said, ‘OK, just leave it and I’ll pick it up in the morning.’”

Still in shock, the next day Basinski used iMovie to layer the footage with ‘d1p 1.1’. The result was what he was to later describe as “the life and death of a melody” soundtracking the aftermath of a world historical disaster. “It was profoundly moving,” he says.

Sequels

The first volume of *The Disintegration Loops*, featuring ‘d1p 1.1’ and ‘d1p 2.1’ (and a still from the video on its cover), was self-released on CD in 2002 and became an instant critical and underground hit. “It just blew up from there,” Basinski says. “My inbox was just suddenly flooded with orders from my website. I was going to the post office twice a day.”

Three subsequent volumes followed in 2003 and from here Basinski built

a catalogue as a prolific composer, in addition to becoming in-demand as a live performer. Then, on the 10th anniversary of 9/11, an orchestral arrangement by Maxim Moston of the first two pieces from *The Disintegration Loops* was performed by the Wordless Music Orchestra at the Metropolitan Museum Of Art in New York.

Moston’s approach was to slowly ‘chip away’ the parts that the musicians played. Additional sounds of decay were provided by the players scrunching pieces of cling film. One written direction on the charts read “popcorn on side of drum skin”. At the end of the performance, the audience listened for a few minutes to complete silence.

Basinski was watching from the audience and remembers, “It was as if everyone had been turned to stone. There were children there, there were old people there. But there wasn’t a sound, until this plane flew over, picking up the F note or something, and then everyone just went nuts. It was really something.”

Photo: Jeremy DeVine

In 2012 came the first vinyl box-set version of *The Disintegration Loops*, released on Brooklyn label Temporary Residence Ltd, whose founder Jeremy DeVine approached a doubting Basinski, and then flew out to the composer’s new home in Los Angeles to convince him.

“When Jeremy came to see me in California, wanting to do the LP box set,” he says, “I was like, Well, y’know, [*the tracks*] are so long. He’s like, ‘Well, classical records... Hello.’ I’m like, ‘Oh yeah, OK, why not? Go ahead and do it.’”

Since then, the orchestrated version of the album has been performed live internationally. “It jumped to pencil and paper and live, lifetime-dedicated and trained musicians, playing what seems like this very simple thing,” Basinski marvels. “But it’s actually quite difficult to pull off, because you have to count a lot.”

“I heard from a wonderful violinist, when we did this incredible outdoor concert Anohni invited us to do in Amsterdam a couple of years ago, that actually

they're sitting there with their arm [stiffly positioned] right here the whole time, and it's very tiring. They're not, like, flying around and getting through everything they usually do, so it's quite difficult.

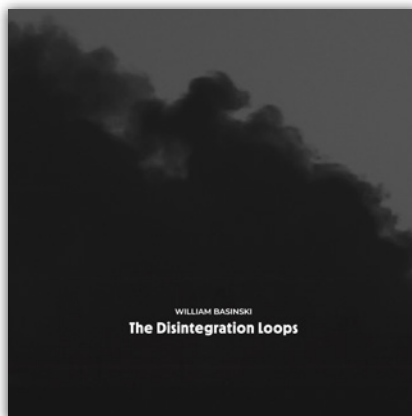
"But every time we've done it, it's just been incredible. And I always cry and thank everyone. This is something I get to go and listen to and take a bow at the end. So it's kind of nice."

Afterwards

Since *The Disintegration Loops*, William Basinski has continued to work with unusual source material. On 2019's *Time Out Of Time*, for instance, he created two electronic pieces using the sound of gravitational waves from outer space recorded in 2015 by the two LIGO (Laser Interferometer Gravitational-wave Observatory) research facilities in Hanford, Washington and Livingston, Louisiana.

"They were picked up on the Earth in these installations they spent years developing to see if they could detect them," he says. "These scientists had been trying to prove the existence of these things, and they finally caught it. This was a rift in space time caused by two massive black holes colliding in space, something like three billion light years away, sending out the most powerful shockwave anyone's ever discovered."

Basinski became involved after meeting LIGO's Jamie Rollins at a party. "Jamie was a huge fan, and he said, 'Hey, we have these weird recordings. Would you like to



do something with them?" And I said, 'Yeah, send them over. I'll see what I can do.'

"I started with some of the sounds of just like clicks, but then also these very low-frequency things. I mean, they had to bring the frequencies up, I don't know how many times, just to be able to hear them.

"I just imagined these waves travelling throughout space time and changing and morphing and moving throughout the galaxies. Then at the end, when they come to Earth, I get very romantic, and it sort of resolves. To me it was like a Gothic love story of two black holes."

A year later, on 2020's *Lamentations*, made and released during the pandemic, Basinski returned to his old '80s tape loops. "A lot of the loops that I used were sort of rejects," he says. "They weren't pretty, and they got set aside in a little takeout container. Then everything had gotten so weird and everything, I thought, Well, let

me see what's in here. Then I ended up coming up with the titles."

In addition, he also — as a result of the success of *The Disintegration Loops* — used old reel-to-reels that had been sent to him by fans. Often, these tapes featured compiled 'playlists' of unidentified music recorded by their previous owners.

"People always send me these giant reel-to-reels," he says. "People's sometimes labelled or usually not labelled playlists. So that's my tape stock [laughs]."

"When I make these loops, I'll record something on one side of a used tape. You don't know what's on it. You just cut a piece, tape it together, and record what you want on it and then I'll turn it over and see what's on the other side. And sometimes you get a great surprise. Like those soprano opera snippets and stuff were just like, Wow."

Of course, these days, tape-styled sound-corrupting digital plug-ins are very popular. Basinski isn't a fan. "I'm an old dog," he points out. "I don't learn new tricks. They are what they are, but it's all fake. I don't try them, I don't use them. I can get it the hard way, honey. Through time... a lot of time."

Incredibly, all these years after realising that his tapes were crumbling during the making of *The Disintegration Loops*, William Basinski has yet to clean out their accumulated dust from the compartment behind the transport on his Revox.

"Yeah," he laughs, "it's still in there." ■■■
The Disintegration Loops – Arcadia Archive Edition is released on 7th November by Temporary Residence Ltd.



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Toontrack 12-String Bass EBX EZbass Expansion

★★★★★



The EZbass EBX expansion line does a pretty good job of covering the most obvious four-, five- and six-string electric bass options, but Toontrack do like to throw in the occasional curveball. The latest addition, 12-String Bass EBX, is just that. Electric basses with 12 strings are, of course, a bit of a niche thing. Mirroring the concept of a 12-string guitar, they tend to come in two physical formats: either a six-string bass with six octave strings, or a four-string bass with two pairs of octave strings. Toontrack have gone down the latter route here and used a Schecter model as the sound source for this library.

From a technical perspective, the EBX follows Toontrack's conventional format. As such, you get the sample library (around 750MB in size) plus a collection of MIDI grooves spanning a range of musical styles, tempos and time signatures and played with the 12-string instrument in mind. The supplied presets include a DI as well as a diverse set of options that provide mix-ready sounds. These provide various flavours of sounds featuring amp and effects combinations and go from just a touch of grit through to full-on, space-filling stereo processing. As well as the usual compact set of controls under the EZbass Effects panel, each preset also include the option to blend in a layer of FM synthesis to add some additional weight to the overall sound.

From the first MIDI groove I passed through the DI preset, it was pretty clear that the sound is something rather special. While the low end is very solid indeed (and can be ably enhanced with the FM synth layer), the additional octave strings deliver both fundamentals and harmonics that place at least part of the sound within territory often occupied by conventional guitars. It's a fascinating — and very appealing — sound that opens up all sorts of new and interesting options, but with the bass able to occupy a wider frequency range, how you voice any other instruments requires some adjustment. It's a lot of fun to explore though, and can change the bass/guitar interaction in some very inspiring ways.

While the 12-string bass has some pretty well-known exponents (for example, Tom Peterson of Cheap Trick; Jeff Ament, who

famously used a 12-string bass on Pearl Jam's 'Jeremy'; and Dug Pinnick of King's X, who has a signature Schecter 12-string), as an instrument it's still something of a rare beast. Albeit in a virtual format, I suspect Toontrack are about to put its distinct sound into the hands of more people than have ever played the real thing. 12-String Bass EBX sounds truly epic and, whether clean or driven, it brings a whole new dimension to the role electric bass can play within a mix. Well worth auditioning for every EZbass owner. *John Walden*

\$89
www.toontrack.com

Sonora Cinematic Transfigured Orchestra Vol 2

Soundbox Expansion

★★★★★



Transfigured Orchestra Vol 2 is an expansion pack designed to run on AudioModern's Soundbox platform (downloadable free of charge for macOS, Windows and iPad). Vol 1 of Transfigured Orchestra featured 30 patches, focusing on the softer side of the orchestral sound palette and including pianissimo and sordino strings, soft French horns and clarinets, organic-sounding woodwind and granular transformations. Sonora Cinematic Transfigured Orchestra Vol 2 delivers what are termed Hybrid Gestures, which I'd describe as cinematic-style orchestral sounds, some of which have been treated using further processing or created from hybrids of other sounds. The pack includes 20 presets, the ones described as Waves offering unprocessed orchestral crescendos. Traditional orchestral instruments used to create the presets include violins, violas, cello, basses, flutes and even a flugelhorn.

The whole library only runs to 750MB, but don't let the small size and low price fool you, as the sounds on offer are impressively cinematic in scope and ready to slot directly into sound-for-picture, games or ambient music. Many of the sounds embody

Sample Libraries

a sense of movement and there's a choice of orchestral crescendos, hybrid ambient washes, or combination sounds that have clearly been crafted with great care. Cinematic Rumbling or Crusher, for example, serve up the familiar slow-attack, ominous wall of basses that preclude some form of doom while the flugelhorn would fit right in with a historical epic where something involving swords and bad attitudes is about to kick off. By contrast the Granular Flutes and This Is Cinema patches are a touch mysterious. The various string sounds also offer the kind of textural movement that many straightforward samples miss out.

Over and above the included 20

presets, it is trivially easy to populate any unused sample slots with samples from any of the packs you already have installed to create new layered patches, and there are some free packs out there that work well in conjunction with these orchestral offerings.

I found some lovely combinations by combining samples from this pack with some taken from Aethervox. This isn't one of those orchestral packs where you pick an instrument and then get a choice of articulations as that's not what the Soundbox platform is designed for, but what you do get are deeply expressive sounds that can fit directly into a film or music score with little or no further adjustment or processing. The pack is not about strict orchestral realism and doesn't cover the entire orchestral instrument range, but instead offers an expressive sense of movement and mood setting. Compact, but genuinely impressive. *Paul White*

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Crow Hill Company Pocket Strings Plug-in Instrument

★★★★★

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» price of \$10 and placing only very modest demands on system resources, the idea here is to provide a portable string instrument which packs a creative punch, both as an excellent sketching tool and a colourful addition to existing libraries.



The clear and attractive GUI provides several immediate sound-sculpting options, allowing users to quickly morph between traditional cinematic-style strings and full-on string-synth textures with a surprisingly large amount of real-time modulation and timbral undulation at one's fingertips.

Looking from left to right on the interface, one first notices two interlocking wheels. The larger of the two wheels is a Timbre dial, allowing the user to crossfade between different recordings of the ensemble playing the same articulation. The inner circle is a standard mod-wheel-driven expression control. Played together this allows the composer to achieve an impressive amount of dynamic expression.

Beneath the wheels we see Glass and Chamber settings. The latter allows for the selection of several different reverb options, from closer-sounding 'chamber' settings, to a nice hall, and a wonderfully cinematic 'Splosh' preset. The innovative Glass setting allows composers to deal with the frequency peaking that can occur when many samples are layered. Turning up the Glass dial applies compression and control to the fundamental frequencies, boosting or reducing them as necessary, while also giving users another fun morphing tool. After this is a digital-sounding, but still rich, Glitten setting, which creates a fast echo with a latent shimmer.

Moving right on the interface, three buttons allow users to immediately switch between long, short and plucked version of their sound. The shorts are punchy and clear and seem to have a round robin selection occurring, while the nice plucks also sneak in a healthy Bartók pizz with a high-velocity input.

The instrument's real creative power comes in the central mixer, which allows users to morph in real time between the traditional chamber string recording, to the classic Solina string emulator (bottom left, marked as 'Vintage Synth'), and finally to the Yamaha CS-80 (marked as 'Epic Analogue'). Crossfading between these three in

sustained passages or in more punchy short driven lines is a real joy.

On the bottom right is a Bass switch, which simply doubles each note with a bass player an octave beneath it (as high as the bass will go, or C4). Beneath that are three large 'shape' buttons, which apply gradual

dynamic changes across a duration of one to four bars. Used in combination with the expression and timbre wheels, dynamically nuanced performances are possible. My only requests would be to maybe include a few more shape types, and to fix the GUI, which currently requires users to click directly onto the shape graphic while the box around it remains inactive.

For only \$10, this is a compact yet powerful little instrument which packs a lot of fun under the hood. It's a pretty little thing with surprising depth, and ably fulfils its role as a small sketching tool and an intriguing addition to your existing string palette.

Mark Nowakowski

\$10

www.thecrowhillcompany.com

Soundpaint Vivaldi's Solo Violin Soundpaint Instrument

★★★★★

Following the release of Disco Studio Strings, Soundpaint return with Vivaldi's Solo Violin, a refined and expressive violin library that brings the essence of Baroque string writing into the modern producer's toolkit. Inspired by the energy and precision of Antonio Vivaldi's music, this library focuses on what makes chamber strings so emotional: clarity, movement and a strong sense of character.

The tone is raw, dry and detailed, less about concert hall grandeur, more about the intimacy of a recording studio. It delivers that timeless, close sound that feels at home in a Baroque sonata but also blends naturally into modern pop, film scoring, jazz or even hip-hop (think of the elegant, melodic string textures in Coldplay's 'Viva La Vida').

If you've used Disco Studio Strings, Vivaldi's Solo Violin will feel like a natural extension. It was recorded in the same room, with the same

players and microphones, and was designed intentionally to complement its predecessor. Where Disco Strings brings lush rhythm and groove, Vivaldi focuses on finesse, articulation and the subtle expressiveness of smaller ensembles.

Technically, the recordings are excellent. The sessions were captured with a Neumann U47, AKG C414 and Royer 121-S, along with four additional spot mics that add depth and flexibility. This gives you total control over the space: close and dry for intimacy, or slightly roomy with natural reflections for more dimension.

The first impression is how real it sounds. Slight detunes, bow noises, vibratos and fast transitions bring an authenticity that feels alive. Everything runs on the Soundpaint engine, known for its expressive response and fast loading. You can shift from lyrical legato to sharp rhythmic phrases seamlessly.

The articulation list is comprehensive and well thought out, and includes Arps & Figures (a library of musical phrases and motifs that cover a wide range of chord types and voicings); Legatos (both mono and polyphonic); Ostinatos & Runs (for movement and energy); Shorts & Sustains; and finally a SoundLab section for creative,

textural experiments.

The Arps & Figures are particularly impressive as they're not static loops. With the built-in Time control, you can adjust the tempo

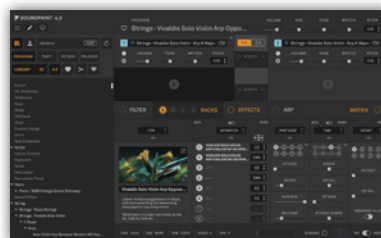
of each motif, while the ADSR envelope controls let you shape how naturally each phrase releases. The Legato patches are another highlight: smooth, expressive and effortless to play, whether used for solos or layered ensembles.

Overall, Vivaldi's Solo Violin is an elegant, intimate and remarkably realistic library that feels instantly musical. When paired with Disco Studio Strings, it forms one of the most versatile and cohesive string collections in the Soundpaint line-up. At \$99, it offers outstanding sound quality and value for any modern composer or producer seeking realistic violins, both solo and ensemble. *Benjamin Boukris*

\$99

www.soundpaint.com

Audio examples of this month's libraries are available at www.soundonsound.com.



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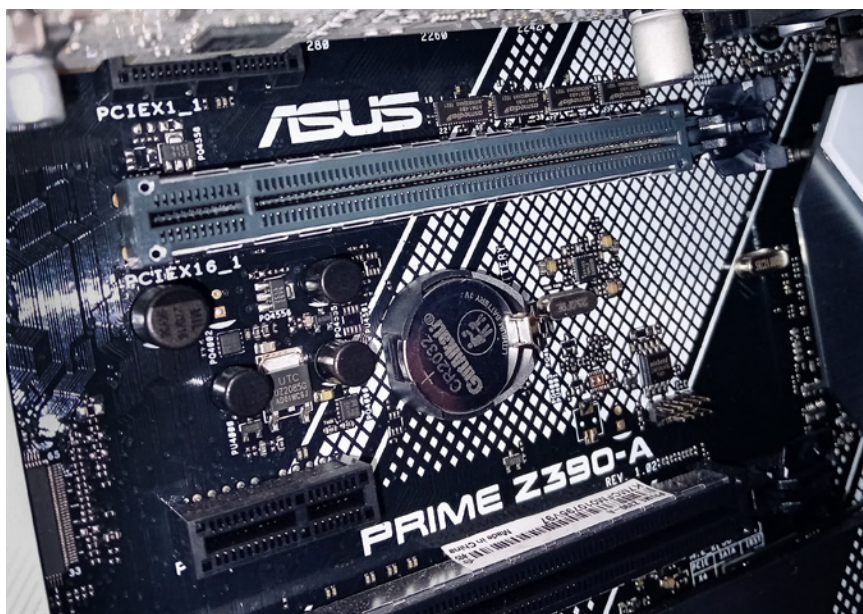
Q Why is my music PC booting to the BIOS?

I have a tower PC that I had built specially for music production a few years ago. It used to work fine, but now I have to boot it twice before I can get Windows running. The first time, it goes to the BIOS, which I don't really understand that well. I don't need to make any changes in there, but if I just save and exit then the machine reboots and I'm taken to the usual Windows log-in screen. It's frustrating because I usually just switch everything on and then go make a coffee, but I have to hang around! What's going on and how can I fix it?

Jason Taylor, via email

Matt Houghton, Reviews Editor

Since your computer's otherwise working fine and it's a desktop/tower machine rather than a laptop (ie. a machine with a rechargeable battery), I strongly suspect that the problem lies in a dead battery on your motherboard. These always fail eventually, and that failure causes precisely the sort of issues you describe — and while you'd be unlucky for it to fail after only "a few years" it's not



Desktop machines usually have a CR2032 battery on the motherboard — accessing and replacing it should be very easy.

impossible. Don't worry, they're designed to be easy to access and replace. Just pop the cover off the case to access the motherboard and you should see the battery — it's usually a small coin-sized CR2032 type. As always, be sure to ground yourself before attempting to

touch anything inside the computer (there are cheap grounding wrist straps to make this easy). Then just lever out the battery and slip the replacement in. If my suspicions are correct, this should get your machine booting up properly first time again.



A bad case of 'the crusties' on a delay pedal that doesn't see enough use!

Q How can I clean 'battery crust' off my gear?

I've had this a few of times now! I've forgotten to remove the batteries I put in some gear — a couple of stompboxes, an Ebow — and then, out of sight, out of mind, I left them like that for a couple of years. And of course, when I go to use them again, it turns out that the batteries have leaked all over the battery compartment and on replacing the batteries and scratching away the 'crust' on the contacts, the gear still doesn't work. I assume I need to do a better job of cleaning away all that detritus, but is there a 'right' way to do it?

Marc Williams, via email

Matt Houghton

Yeah, we've all been there at one time or... many others! Our Technical Editor might well tut, as I know he's rather more careful and organised with his gear than I am with mine, but I've had this happen with pedals, handheld recorders, my Ebow and even a mic. But thankfully I haven't yet discovered anything I couldn't bring back from the dead. As you say, the key is cleaning.

First, you need to remove the offending batteries, and wipe/scrape away any goo and loose dust left over from the leak. Then dispose of the batteries safely. (Just in case, don't do any of this with bare fingers!) As for cleaning the gear, the specifics are going to depend somewhat on what the gear is and the access you have to the battery compartment. But ideally you want to bathe the compartment in something acidic to neutralise the leak and to dissolve all the 'crud'. I use a weak (circa 6%) distilled >>

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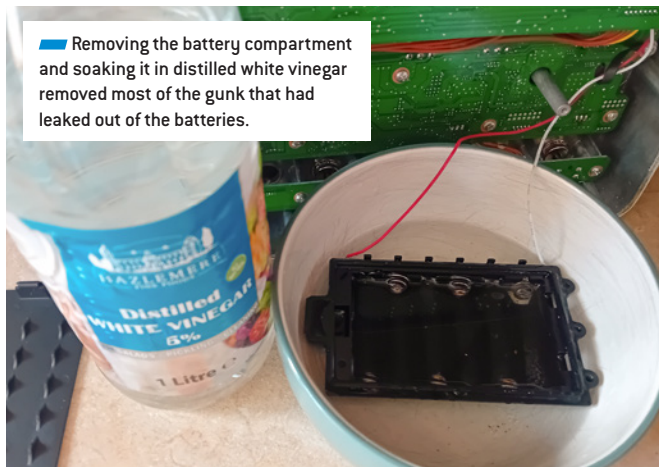
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■ Removing the battery compartment and soaking it in distilled white vinegar removed most of the gunk that had leaked out of the batteries.



■ A final dry, scrape, and squirt of contact cleaner finishes the job.

» white vinegar solution for this, which is very affordable and doesn't leave behind any nasty residue.

The vinegar will do most of the work, but you might find on some stubborn areas you need to mechanically scrape or rub the last bits away. And if the practicalities of the device mean you can't actually bathe it, then you can apply it with cotton wool and rub it in. With a bad leak, the damage may extend beyond the compartment itself, so check for that — use the same method to address that, though note that if the batteries leaked onto any components or PCB it may well be unsalvageable.

When done, rinse and dry the area (I use kitchen towel for this). Then you just need to make sure the battery contacts are clean enough to make a good contact. Again, you can just give these a good scrape or rub with something abrasive and blow away any dust, and then, for good measure, give them a squirt of contact cleaner such as DeOxit D5 or Kontakt Super 10. That should be job done: stick some new batteries in there and give your gear a try. If that hasn't done it, it's time to let go, buy a replacement, or if the gear's rare or valuable maybe contact an electronics repair specialist.

Q How much do I really need to spend on an SD card for my digital audio recorder?

I'm about to flex my credit card and get a Tascam Studio Bridge, but I need an SD card for it. I already have a couple of cards lying around, but I'm not sure if they'll be up to the job. The price of SD cards seems to vary a lot now, so what should I be buying to get the best performance without wasting money? The manual does give some information, but I'm finding the choice and price differences a bit overwhelming!

Simon Walker, via email

Matt Houghton

Tascam specify "SD cards that are Class 10 or higher and compatible with SD, SDHC or SDXC standards". SD is the oldest type, and can store up to 2GB — not a lot, when you're recording 24-track audio. SDHC came next (in 2006), and its last two letters stand for High Capacity, denoting that this format was capable of storing up to 32GB. Finally SDXC (Extended Capacity) cards could give us 64GB to 1TB of storage (the Tascam can support only up to 512GB at the time of writing). In my review tests



of the Studio Bridge (<https://sosm.ag/tascam-studio-bridge>), even an SDHC card I bought about 15 years ago let me record 24 channels at 16-bit/44.1kHz. An only slightly more recent card worked flawlessly for me at all word lengths and sample rates. So, if you have a card or two lying around already, one of them is likely enough to test your new toy. But I'd probably still buy a dedicated card, not least because if it lives in the Studio Bridge rather than being shared with other devices, you'll never wonder where it is!

An SDXC type makes sense, as 128GB or 256GB cards aren't that expensive. Class 10 was introduced in 2009, and most cards will now meet that criterion. Each card displays other information on the front, notably the maximum achievable read speed. That might be great for marketing, but we're interested in the sustained write speed, and even if that's not printed on the card, there will be clues.

First, there's a number sitting inside a 'U' icon. With modern cards, this will likely be a 1 or 3, and the U1/U3 designation tells you that the card has either a UHS-I or UHS-II bus. This means write speeds of *up to* 104 and 312 MB/s, respectively. But we're interested in the *minimum* write speed. A V rating (V30, V60 or V90) denotes the speed, and any with one of those ratings will do the job. High-capacity V60s and V90s are where things start to get really expensive, but they're overkill for this application. An SDXC card with a V30 write speed rating will be ample — for instance, I'm currently using a couple of 128GB Lexar Silver Plus V30 cards (about \$23 each) for my Studio Bridge and they seem to work very well. ■■■

NEXT MONTH IN **SOUND ON SOUND**

Dave Stewart

"I don't make any demos, and never have. If I had something on the Portastudio, it was bounced onto the eight-track and that was part of the master."

The legendary songwriter and producer explains how Eurythmics' hits were made — and why he's recorded an album of Dylan covers on his iPhone.

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A.P.

EURYTHMICS

AVE Anderson

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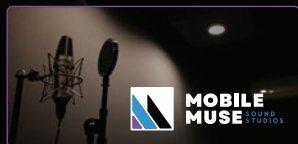
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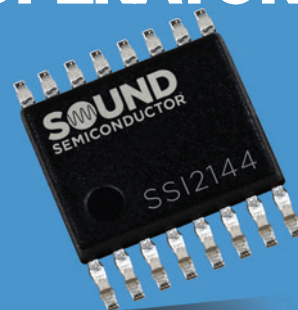
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S/O/S WHY I LOVE... SMEM

SIMON HORROCKS

From multifunctional army knives to chocolate, watches, banking, yodelling and (surprisingly) garlic presses, even the most exhaustive Python-esque “What have the Swiss ever done for us?” list would be unlikely to include synth-related stuff. Neighbouring France (Arturia, Daft Punk), Germany (Behringer, Kraftwerk), Italy (IK Multimedia) and even Austria (Falco) all have more well-known links to electronic music than the inconspicuous town of Fribourg, nestled in a landlocked country of lakes and mountains.

But even if you get there, the guidebooks won't help, and good luck finding the small sign in a huge industrial estate, before going down three floors in a lift, to a dust-free, humidity-controlled underground man-cave — aka the Swiss Museum of Electronic Instruments, the synth wonder of the world.

My guide was Vic Genna, well versed in the art of watching people's jaws drop as he narrates you through the floor-to-ceiling shelves stacked with over 5000 historic synths, drum machines, keytars, toys, pedals and computers. The kit is in immaculate condition too.

“As well as all the synths,” he explains, “we have a huge number of older organs. These are bulkier in terms of how much room they take up in our space. Although some of them connect to the birth of synthesis, such as the 1940s-built Hammond Novachord, which was in fact technically the first polyphonic synth and has eight oscillators. The future of the rest of the organs is uncertain, but staff and volunteers here all have a say what happens to the collection, what projects we get involved with, what becomes collectible, what doesn't. It's extremely democratic.”

Around 95 percent of the SMEM collection was founded by Basel collector Klemens Nikolaus Trenkle in the 1980s — he clearly bought

everything at an advantageous time price-wise, before acid house, dance music culture and later DAWless jamming inflated the prices. Working partly as a musical instrument dealer, in 2016 he co-founded the SMEM ‘living museum’ and cultural concept.

You see, SMEM isn't just a heap of dusty relics in glass boxes. All the gear is lovingly restored, catalogued, learnt, used in collaborative projects and incorporated into established workshops and artist residencies. Nothing encapsulates this more than the SMEM playroom, a living workshop and studio space separate to the larger store, where people can get hands on, experiment, record and collaborate.

The whole place is nothing short of miraculous, and unique in Europe. Most if not all of the big manufacturers (Roland, Korg, Yamaha and all the others) are apparently as jealous of this set up as you or I. Multiple versions of 808s, Wasps, Moogs, Prophets, and things you've never seen before from countries you've

never visited are hanging around waiting for you to gawp at, touch, feel.

If you think about the hardware versus DAW history of electronic music, you'll realise that this place is like an infinite VST library. Except every app/box here is the original physical artefact, with a fascinating authentic sound and haptic user interface that would teach modern app designers a thing or two.

But apart from all that, why do I love it? Well, you know how lovely, kind and well-spirited all those people you've ever met who love synths and electronic music are? This is where they gather. This isn't about superGAS, it's a spiritual church of musical synthesis and a tribute to its creators — a sacred space for people who worship the god of synth to gather and celebrate; to live and breathe its future, present and past all in one go.

Yeah, all of that, in the country that brought you cheese with holes, and Heidi.

In fact, the Swiss love of craftsmanship, manufacturing, design, functionality, and most of all generosity of spirit, makes Switzerland the perfect home for such a thing. As awe-inspiring and inspirational as the mountains and lakes themselves. ■■■

Photo: Simon Horrocks

“SMEM isn't just a heap of dusty relics in glass boxes. All the gear is lovingly restored, catalogued, learnt, used in collaborative projects and incorporated into established workshops and artist residencies.”



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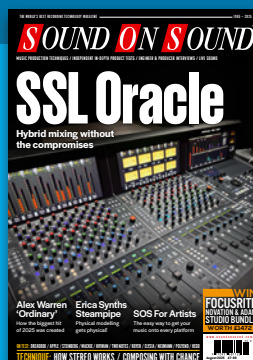
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SOUND ON SOUND

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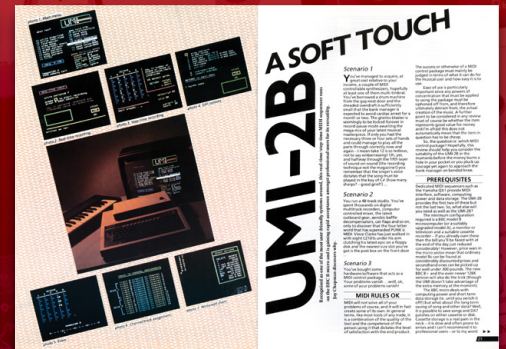
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