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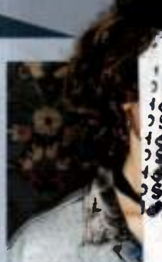

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Editor's Note

Stream of consciousness

When it comes to the ways in which we listen to and engage with music, it's easy to be dismissive of new technology. After all, we all tend to look back fondly on whatever format we grew up with, whether that be vinyl records, cassette tapes, physical CDs or some form of more recent technological development.

As I write this we've just passed the 20th anniversary of the iPod, and I even find myself getting all nostalgic for my first portable MP3 player. To be fair, after years of loading up a limited volume of songs onto tapes, CDs and MiniDiscs – and dealing with the associated skipping, warped tapes and other audible anomalies – suddenly having access to that amount of music in the palm of your hand was pretty incredible.

In the eyes of some musicians, producers and music fans, the rise of streaming is the worst thing to happen to music in decades. There are, undoubtedly, some downsides. For one thing, streaming royalties tend to be pretty poor for most artists, making it harder for many up-and-coming acts to make a living from their music. Having such instant access to music also, you could argue, removes some of the fun of digging for rare finds and hunting obscure releases. For music fans though, that's surely offset by having instant access to an endless catalog to explore.

From a production point of view, as we explore in this issue's masterclass, the rise of streaming has had the positive effect of ending the 'loudness wars' that dominated the '00s, destroying all dynamic range in their wake. Whatever your thoughts on streaming, it's a reality we have to deal with, and our masterclass is here to help you get your tracks ready.

We hope you enjoy the issue.

SI TRUSS
EDITOR-IN-CHIEF

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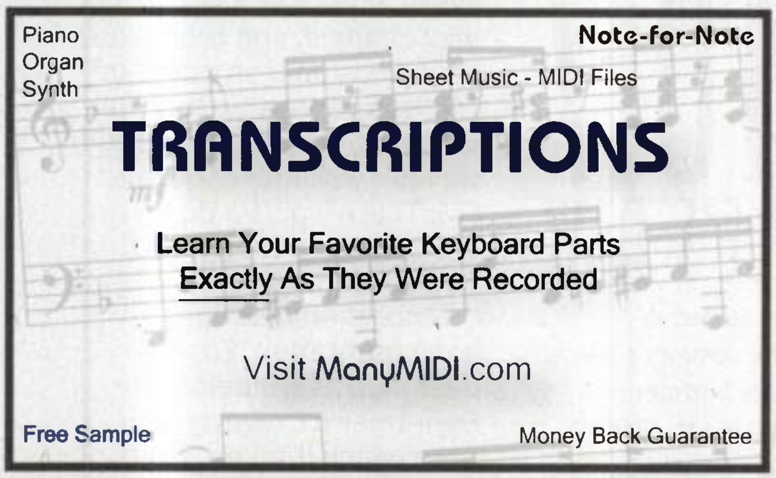
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NEW GEAR

LOGIC PRO 10.7

BRINGS DOLBY ATMOS TO APPLE'S DAW

Apple has made good on its promise to bring spatial audio authoring tools to Logic Pro X. Anyone who downloads the version 10.7 update can now create tracks that support the Dolby Atmos-powered format, and then release them on Apple Music if they wish to.

New mixer and panner controls have been added to Logic Pro to enables users to access Dolby Atmos-compatible surround channels, and 12 existing Logic plugins (including Space Designer, Limiter, Loudness Meter, and Tremolo) are now Spacial Audio-friendly, too.

The 10.7 update also adds the Producer Packs that landed in GarageBand over the summer – users can now access beats and samples from the likes of Boys Noize, Mark Lettieri, Mark Ronson, Oak Felder, Soulection, Take A Daytrip, Tom Misch, and TRAKGIRL. In fact, you get 2,800 new loops, 50 new kits, and 120 new patches, along with the original multitrack project of Lil Nas X's Montero (Call Me by Your Name), which includes a Dolby Atmos spatial audio mix of the track.

Logic Pro 10.7 is available now as a free update for existing users. The price for new customers is $200.

BEHRINGER TAKES ON THE MOOG DFAM WITH THE EDGE SEMI-MODULAR PERCUSSION SYNTH

Behringer has been a little quiet of late, but it's just come bursting back into the spotlight with the announcement of the Edge semi-modular percussion synthesizer.

Already being likened by many to the Moog DFAM – which also happens to be a semi-modular percussion synth – this has a striking pink chassis, and promises a sound "as bold as its looks".

This is generated by dual VCOs with pulse and triangle waves, oscillator sync, and FM. There's also a dual 8-step sequencer, 15×10 patch matrix, and "comprehensive" MIDI implementation. Edge is designed as a sidekick for Behringer's Crave synth, from which it takes plenty of design cues. It has a list price of $219, but we don't yet know when it is set to be released.

IZOTOPE RELEASES RX 9

iZotope's RX software has always been a godsend to anyone who needs to improve imperfect audio, and now, with the release of version 9, it's just got even better.

The focus with this update is on tackling "emerging audio capture and production issues to restore damaged, noisy audio to pristine condition," which means that some of the most popular RX modules have been overhauled, particularly those that are widely used in post production.

Indeed, the flagship RX 9 Advanced is specifically targeted at post production professionals, with the new version of the Dialogue Isolate module making it easier than ever to extract clean dialogue from its environment without artifacts.

Elsewhere, the new Complex mode in Ambience Match is designed to connect dialogue and ADR cuts with real background movement and textures, with multichannel support up to Dolby Atmos 7.1.2.

Common to both RX 9 Standard and Advanced is the new Dynamic Mode in the De-hum module, which promises to remove any amount of hum and interference in a single pass. There's also 30-step undo in the History list, and a Restore feature that enables you to roll back an audio selection to any previous step in said list.

"For RX 9, we have revisited some of our most important processing tools and made them even better," says iZotope's Principal Product Manager for RX, Mike Rozett.

"We wanted to focus on the fact that dialog is getting noisier and noisier: from location shoots to warehouse sets, to wireless interference on sound stages, to ADR that's being recorded remotely in cars and closets and kitchens instead of in studios. The industry is facing more and more noise, with less time to fix it. We're here to help."

iZotope RX 9 is available now for PC and Mac. The Standard version can currently be purchased for $299 (regular price $399) while the Advanced version can be yours for $799 (regular price $1,199). The software is also included in the RX Post Production Suite 6, which is on offer at $999.

SPECTRASONICS

ADD FOUR NEW THEMED INSTRUMENTS FOR OMNISPHERE 2

It's not Omnisphere 3 – there's no word on if or when that will be released – but Spectrasonics has unveiled its new Sonic Extensions, add-ons for Omnisphere 2 that grant users new sounds and features.

At launch, there are four Sonic Extensions: Undercurrent, which is designed for "dark electronic scoring"; Nylon Sky, an ambient acoustic guitar; the retro-sounding Unclean Machine; and Seismic Shock, which is said to be suitable for heavy, modern electronica.

Each of these comes with not only a deep, multi-gigabyte set of multisampled sounds, but also two new and exclusive effects. Once you have the Sonic Extension, these effects can also be applied to all of your Omnisphere content, and also to 'satellite' Spectrasonics instruments such as Keyscape and Trilian.

What's more, each Sonic Extension has its own bespoke control set, so it looks like an instrument in its own right. You'll need to have Omnisphere 2.8 or later installed if you want to run any of them, though.

The Sonic Extensions are priced at $149 each, but there are savings to be had if you buy in bulk. If you buy two at once you get 20% off, while purchasing three at once gets you 30% off. Unfortunately, it seems that these discounts can't be applied retroactively, so you won't save anything if you buy one Extension now and another at a later date.

ROLAND'S SP-404MKII PROMISES NEXT-LEVEL PERFORMANCE SAMPLING

Following sieve-like levels of leaking in the run-up to its launch, Roland has 'dropped' the SP-404MKII, a new version of its performance-focused portable sampler.

With a design based on both user feedback and Roland's own research, this promises to be the fastest and best SP sampler yet.

This MKII version of the SP-404 promises everything that users loved about its predecessor, but also new features such as more expressive pads and updated knobs. Boot time has been speeded up, as has project loading and sample import.

The SP-404MKII comes with 16GB of internal storage, which is loaded with a collection of "curated" samples. Power can be provided by AA batteries or a power bank, and you can interface with mobile devices via USB. Further connectivity includes dual headphone outputs and a mic/guitar input.

The OLED screen is another potential highlight, with a zoomable waveform view promising to ease the process of sample editing. Samples can be chopped up in realtime or you can auto-chop, with envelope and pitch shift enabling further tweaking.

There's a new resampling workflow, too - you can now re-record patterns and effects layers. Skip Back Sampling, meanwhile, enables you to capture the last 25 seconds of audio from your most recent performance, ensuring that those unforeseen moments of inspiration won't be lost.

In terms of effects, you get both your SP favourites - the likes of the Vinyl Simulator and DJFX Looper - and new processors such as Lo-fi, Cassette Simulator, and Resonator. The Vocoder, Auto Pitch, and Guitar Amp Simulator effects, meanwhile, can be applied directly to the mic/guitar input.

Elsewhere, improvements have been made to the sequencing workflow. You can apply adjustable input quantize and shuffle for custom swing, link pads so that you can trigger multiple samples, or hit the Roll button for variable note repeat. The BPM can be set on a per-note basis, enabling instant tempo changes, and sets can be chained together.

Finally, there are new customization options: download a faceplate template and you can create custom overlays, and you can personalize both the logo on the startup screen and your screensaver.

The SP-404MKII will be released in November priced at $500.

PRESONUS'S SECOND-GENERATION R-SERIES MONITORS PROMISE MORE CONTROL AND IMPROVED PERFORMANCE

PreSonus is updating its studio monitor range with two new models: the R65 V2 and R80 V2. These second-generation speakers replace the original R-Series, promising more control and improved sound.

Both the R65 V2 and R80 V2 feature the Acoustic Tuning controls from PreSonus's Eris monitor line. So, you get Low Cutoff, Mid Frequency, and High Frequency controls, along with a three-position Acoustic Space switch that can be used if you have your speakers against a wall or in a corner.

The upgraded 140W power amp, meanwhile, is designed to deliver warmer, smoother frequency response and distortion-free sound, even when you crank up the volume. The tweeter is a custom-designed, 6.8-square-inch Air Motion Transformer (AMT) affair that we're told is great for hearing the ultra-high frequencies that can add 'air' to the sound. This also features a thin folded Kapton membrane that promises to pick out the subtlest details in your music, and we're assured that you can expect a wider listening sweet spot with the AMT design.

Both monitors feature a custom-woven, composite woofer (sized at 6.5 or 8 inches, depending on which model you go for) while balanced XLR/1/4-inch TRS and unbalanced RCA input connections give you plenty of flexibility. The R65 V2 and R80 V2 are available now priced at $330 and $430 respectively (for a single monitor).

UNIVERSAL AUDIO INTRODUCE AFFORDABLE VOLT INTERFACES

Universal Audio's Apollo audio interfaces have become a byword for quality, but for a lot of producers, their relatively high prices put them out of reach. Fear not, though, because UA has now introduced the Volt range – a new line-up of affordable USB audio interfaces that promise "classic studio sound".

It's worth pointing out immediately that none of these interfaces supports the DSP-powered UAD plugins – you'll still need an Apollo if you want to run any of those – but all of the Volts (there are five in total) do include a Vintage Mic Preamp mode.

This is inspired by the mic/line preamp in UA's 610 tube console; tube emulation circuitry is designed to let you dial in "rich, full sound on vocals and instruments".

If you're willing to pay a bit more, you can choose one of the Volt 76 models, which add an analog circuit based on UA's 1176 compressor. With this engaged, users can choose from presets that are designed to add clarity and punch to vocals, guitar and other input sources.

In terms of connectivity and control, simplicity is the watchword as far as the Volts are concerned, with just a few knobs and easily-accessible direct monitoring. There's also 48v phantom power so you can plug in condenser mics, along with MIDI I/O.

Offering an appealing retro aesthetic, the Volts are made of metal and promise to "last for decades"

All of the Volt interfaces offer 24-bit/192kHz audio conversion and run on PC, Mac, iPad and iPhone. What's more, each model entitles you to a music software bundle that features contributions from Ableton, Softube, Celemony, Relab, Plugin Alliance, UJAM and Spitfire Audio.

Prices start at $139 for the 1-in/2-out Volt 1, with the 2-in/2-out Volt 2 costing $189. The compressor-equipped '76' versions of these models cost $249 and $299 respectively.

NEKTAR ANNOUNCES

THE IMPACT LX MINI, ITS MOST POWERFUL MINI MIDI KEYBOARD YET

Promising everything you need to play, perform and produce on the move, Nektar's new Impact LX Mini is a little MIDI keyboard with a deceptively big feature set. The controller is built around 25-note velocity-sensitive keys, with a joystick giving you hands-on pitchbend and modulation control.

The fun stuff can be found up above: there are two independent arpeggiators, eight LED drum pads and eight knobs (plus a volume control). Most controls are MIDI-assignable. The keys and pads have their own arpeggio and note repeat engines, meaning that you can trigger different 'rhythmical figures' with each, and on different MIDI channels if you wish. Parameters for these engines can be adjusted with the knobs.

There's also the Part 2 performance feature, which can be used for momentary setup changes. Press and hold one of the two dedicated buttons and you can instantly transpose the keyboard, switch to another MIDI channel or layer a second sound – release it and you'll return to the original setup.

The Impact LX Mini also offers integration with plenty of popular DAWs - you can control your transport, navigate tracks/projects and open/close DAW and plugin windows. Instrument mode, meanwhile, gives you control of up to 16 parameters per plugin; all assignments can be stored and then recalled.

The included software bundle features the Bitwig 8-Track DAW and the Bitwig Essentials content package. This contains loops, samples and more than 50 software instruments and effects.

The Impact LX Mini will be released in November priced at $120.

Minimal Audio Rift

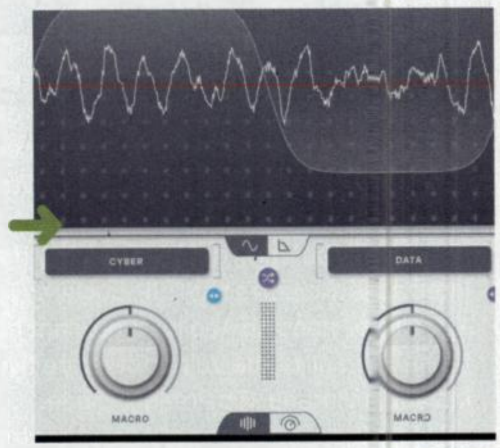
Rift, a recent release from plugin newcomers Minimal Audio, is a cutting-edge hybrid distortion that imparts a modern, unapologetically digital distortion character on your audio signal. It's a plugin with comprehensive processing power, that's accessible and easy to use at the same time, and it's garnering attention for all the right reasons.

Minimal describe Rift as "The world's first bipolar distortion plugin", and it's true that this concept has been largely unexplored until now. The general idea is that the plugin splits an incoming waveform into its positive and negative portions, which can then be mixed and modulated in all sorts of fun ways.

Rift gathers 30 distortion algorithms and organizes them into five categories, starting with soft clipping and moving into heavier ground with bit-crushing and sample-rate destruction. Let's take a look...

1. Play View

If you like to live life in easy mode, Rift's Play View is the place to start... and it's also the place you start when opening up the plugin. We start surfing the presets to see what Rift can do to our sound. In this view, the two main knobs, placed on the left and the right, process only the parts of the incoming waveform that are positive (left knob) and only the parts of it that are negative (right knob). The two Macro knobs on the bottom are pre-assigned in the presets to give you the best chance to mess with the sound quickly.

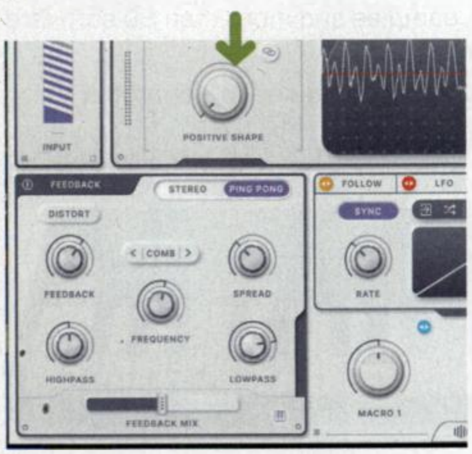


2. The Filter

Advanced view contains many of the same elements as Play View, with all extra modulation controls on top. Switch the Filter on and choose between Pre or Post filtering. You can get some intense results by driving resonance into your distortion using the Pre filter. The filter features 24 algorithms and four filter types: Basic, Morph, Peaking, and Harmonic. A Cutoff parameter can be assigned to track MIDI input quantized to notes and scales, or worked into standard frequency cutoff. There's also Resonance, Morph, and Filter Spread parameters, and a Mix slider to blend the effect with your signal.

3. Feedback

The Feedback panel can be toggled on and off, and lets you return a proportion of the output signal to the input, for a host of stereo and ping-pong delays, distorted feedback, chorus and flanger-like modulation, resonators, and even frequency/note-tuned feedback. You can set the feedback rate in notes (MIDI input or specific notes), Hz (for comb filtering effects), milliseconds, and BPM-synced note divisions. Further shaping options are available under Amount, Distort, and Spread parameters. Added to this you'll find high pass and low pass filters, and a Mix slider.



4. Modulation

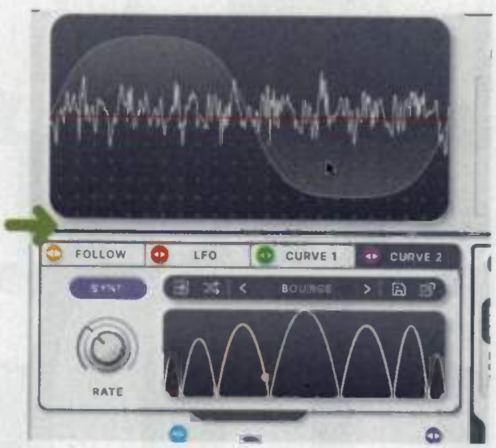
To assign modulation sources to parameters, all you need to do is to drag and drop onto your chosen controller and then slide up or down to select the modulation range. The LFO has a Randomize slider that you can use to introduce organic variations to the modulation. Every time it completes the cycle it redraws the LFO shape. There's plenty of fun to be had here, try assigning the LFO to the Cutoff on the filter and setting the Randomize slider in the middle. Turn off the Sync button and push up the Rate, and you'll see the different wave shapes being created.

A menu of distortion

Rift lets you choose from a menu of 30 distortion algorithms for each of the positive and negative waveform halves. They've been neatly categorized; you can opt for waveshaping, wavefolding, noise, bit depth reduction or sample rate reduction. Although 'bipolar distortion' will be new ground for most, it doesn't feel like it when using Rift, as the large display in the middle clearly shows any changes you're making to either side of the waveform. You can add a substantial amount of drive in the top panel (click the 2x multiplier for ultimate decimation) as well as blend between the two different algorithms in two modes: Hard (brighter, more defined) and Smooth (warmer, less defined).

5. Curve View

Rift's Curve View is another way of visualizing the modulation you apply to the audio signal. In the two Curve modulators, you get over 50 preset curves that range from simple to complex, as well as the ability to edit and tweak to your liking. You can cook up anything from ramps to envelope sequences here, or generate random curves as a useful starting point. Assign Curve 1 to the Spread on the filter and you'll see that the modulation is only going one way – right-click on the green icon beside the knob and you'll have the option to set it in bipolar mode.



6. Pitch Tracking

With powerful pitch tracking capabilities, Rift opens the door for adding pinpointed harmonic content to your sound. The MIDI functionality lets you select either a note or a chord and then arranges the output frequency content accordingly. You could add a hint of A minor dancing around the outskirts of your input signal, or create a chaotic atonal blend with multiple notes. Live MIDI tracking means you could also choose to run a full melody or chord progression through the plugin, and let its various modules clock to your track.

7. Layering Modulation

When you assign a curve to a parameter you can choose a depth mod by right clicking on the green icon beside the control. Choose Curve 2 and you'll see a purple ring appear inside the green one. You can use this to control the modulation that the first curve is actually applying. If you set the Rate quite slow on Curve 2, you can see that the modulation from Curve 1 is only being applied partially when Curve 2 is all the way down. As it comes up the modulation will reduce. If you want to adjust the modulation amount of Curve 1, hold down shift and drag the slider up and down.

EQ & DYNAMICS 101

You don't need the latest and greatest plugins to get a polished-sounding mix. We show you how to get the most from standard EQ and dynamics processors

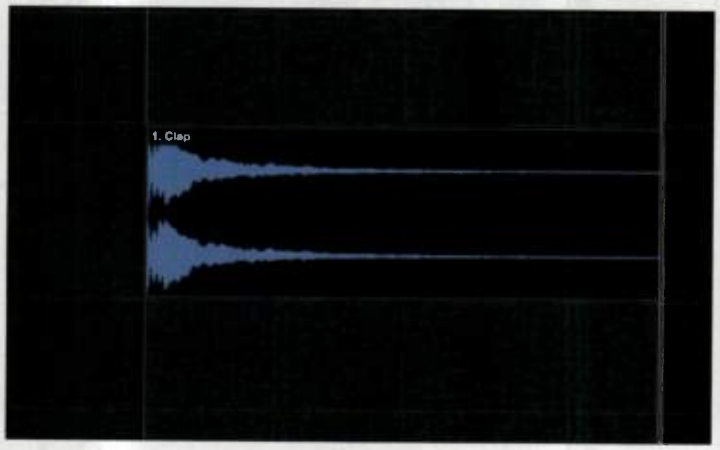
It's a common misconception among new producers that anyone making decent-sounding music must be using hundreds if not thousands of dollars' worth of high-end plugins. The reality is that many artists simply use their DAW's stock mixing effects. The producer who knows how to get the results they want from basic equalizers, compressors and gates is much more likely to make quality mixes than the producer with a ton of expensive plugins but only a shallow understanding of how to use them. In fact, having too many plugins in your arsenal can be counterproductive: those who only have access to limited tools tend to learn to use those tools far more comprehensively. So, rather than spending a fortune on new effects, we recommend mastering the ones you already have to hand in your DAW first. Not only will this help you to make better mixes now, but it'll also enable you to make more informed decisions as to what types of sounds and features you require if and when you do decide to upgrade.

In these walkthroughs, we'll show you how it's possible to get the most out of the most basic EQ and dynamics processors. We'll reveal how some plugins can be used for more tasks than you might imagine: for example, EQ can make it possible to boost the overall level of a signal by attenuating peaks in its frequency spectrum, a compressor can be used to control and enhance transients by turning up its attack time, and a gate can be used to tighten up percussion samples.

By following these guides you'll increase your understanding of these simple but powerful effects, and improve your ability to mold sounds into the shapes you desire. With this knowledge you'll be able to achieve a far greater variety of results than you would relying purely on presets, and you'll get better overall mixdowns as a result.

We're using Cubase, Live and Logic but the processes work in any DAW. Audio examples and files can be found at the link on this issue's contents page.

Step by step 1. Shaping drums with a gate in Cubase

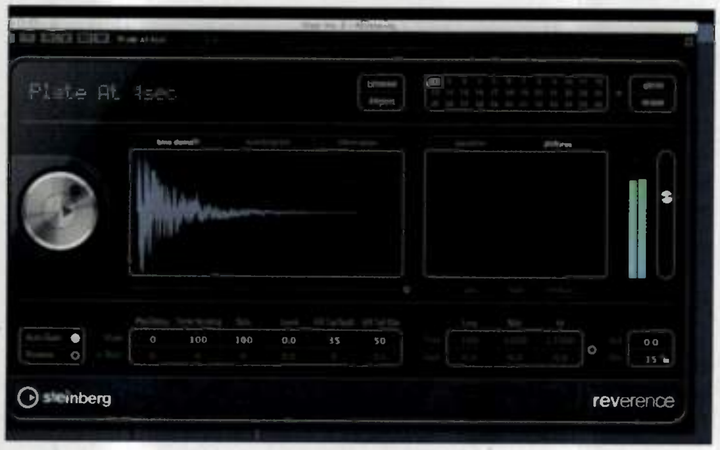


1 Noise gates were originally invented to reduce the overall level of noise in multi-track analog recordings, but they're still useful in the digital age for, amongst other things, shaping drum sounds. Create an audio track in Cubase and drag Clap.wav onto it.

2 This clap has a lot of reverb on it – in fact, it's so long that the sample ends before the reverb tail has finished! This clearly won't do. To fix the issue, click the Insert tab in the Inspector, then the triangular disclosure button on the first slot. Select the Gate effect from the Dynamics folder.

3 Even at its default settings, the Gate gives us a much more natural sound, fading the tail of the reverb to silence. Let's get a bit more aggressive with it. Turn the Release level down to 11. This gives us just a short clap sound. Now it's impossible to tell there was ever a reverb on it in the first place.

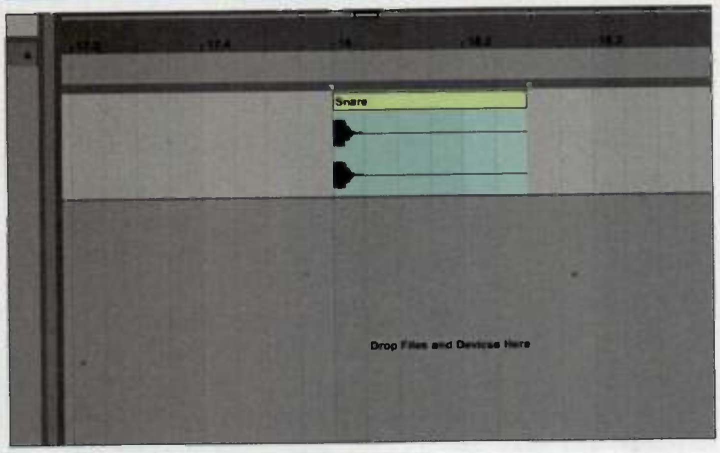
4 You can make the sound even shorter by turning up the Threshold level, if you like. Let's set it to -14dB. Now we've got our shorter clap, there's nothing to stop us adding our own reverb for a different sonic character. Add REVerence after the Gate effect in the Insert strip.



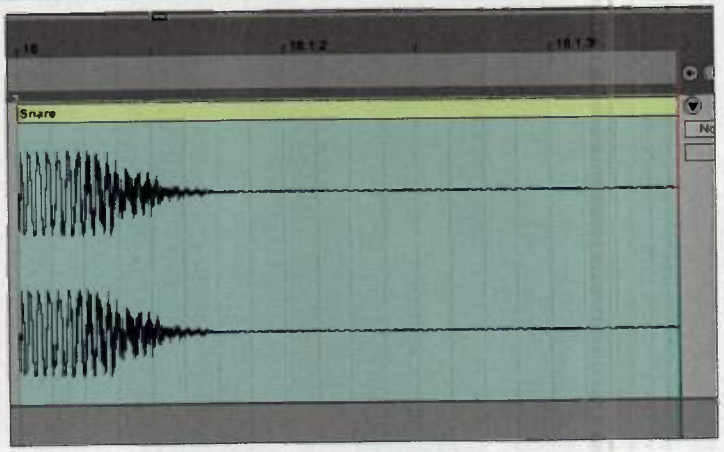
5 Click the empty patch name slot to bring up the list of presets. Select Plate At 4sec and turn REVerance's Mix level down to 15. This reverb is very intense, so it sounds pretty loud, even at this low level.

6 Add another Gate after the reverb, and turn the Release down to 61. Again, this tightens things up, giving us a more usable reverb tail. You can A/B the original and new versions of the clap by clicking the Insert strip's Bypass button. As you can hear, we've completely replaced the character of the original reverb with a different effect.

Step by step 2. Using compression to enhance transients in Ableton Live



1 Compression isn't just for reducing dynamic range and increasing average volume level – it can also be used to enhance the transients of your drum sounds. This is useful when you have a drum sound with the right character, but that's lacking that initial hard attack. Launch Live and drag Snare.wav onto an audio track.

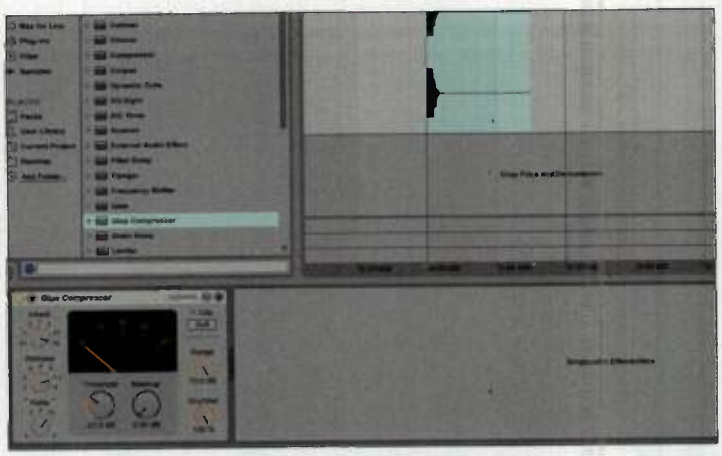


2 As you can see from the snare's waveform, its dynamic range is pretty low. The fat sausage shape at the start means the beat is indeed very loud, but what if you need more impact at the beginning of the sound? Add Glue Compressor to the track.

3 We'll use compression to lower the level of the sample, but only after the transient stage has passed. To do this, turn the Attack parameter up to 30ms. This means that the compression takes 30ms to reach its full effect after the Threshold has been exceeded.

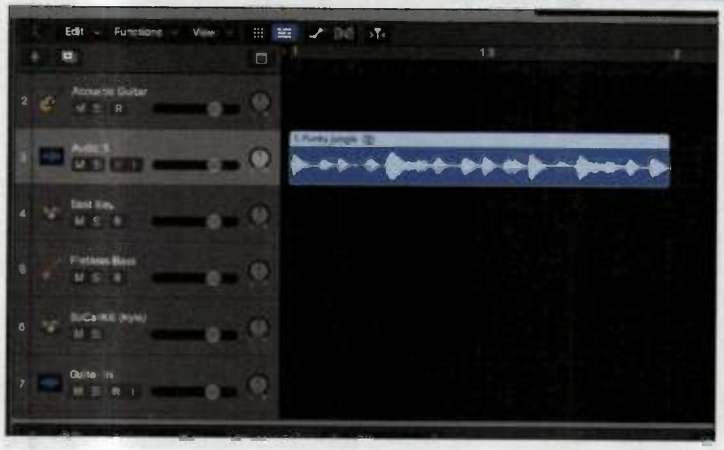
4 Turn the Threshold down to -27dB. On playback, you'll hear that the sound has less perceived volume overall, but the transient at the start of the snare is now much more solid and satisfying.

5 The Release knob determines how long it takes the volume level to return to normal after the signal drops below the Threshold. The longer this is, the longer the compression lasts. Turn it right up to 1.2s to get as tight a sound as possible.

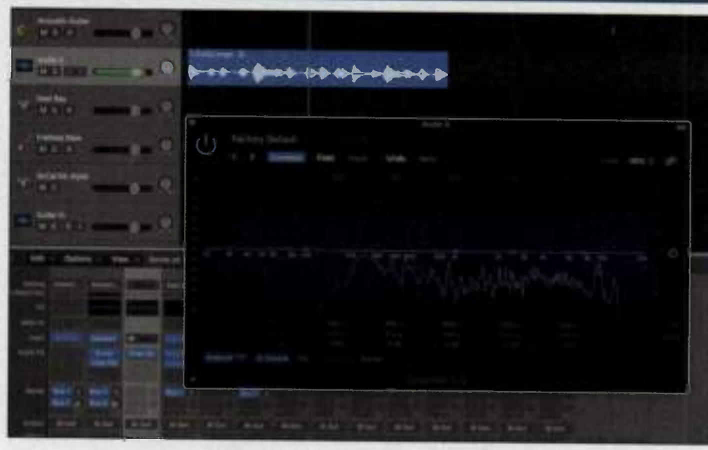


6 Now turn the Ratio up to 10 to give us the most severe compression Glue Compressor is capable of. With everything set up, you can use the Threshold knob to fine-tune the balance between prominent attack and overall loudness.

Step by step 3. EQing beats with spectral analysis in Logic



1 When you're dealing with a sound that has extreme frequency spikes – as some drum loops and breaks do – spectral analysis can help you locate their positions. Create an audio track in Logic and drag Funky jungle.wav onto it. Set Logic's tempo to 140bpm.



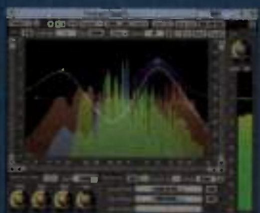
2 Click the Audio FX button on the audio track in the Inspector, and select EQ»Channel EQ»Stereo. Click the Analyzer button on the left-hand side of the Channel EQ's interface, and when you play the project back you'll see the audio signal represented visually behind the EQ curve.

3 Channel EQ's analyzer isn't particularly configurable, so let's use something more suited to the task. Add Voxengo SPAN (free download from voxengo.com) after the Channel EQ, and use the horizontal bar at the bottom of its frequency display to zoom in on the high end of the frequency spectrum.

4 Using SPAN, we can see that the peaks occur at 9.4kHz and 12.2kHz. Return to Channel EQ and set two band EQs with Qs of 2.00 to take off 10dB at 9,400Hz and 12,200Hz. This tames the peaks, giving you more headroom to turn the channel up and make it louder in the mix.

FOUR CLASSIC EQUALIZER PLUGINS

DDMF IEQPro **$37**
This includes a host of great features such as serial and parallel routing options and 19 filter modes (including several Butterworth filters) plus lots of bands and a built-in spectral analyzer. It's also very reasonably priced.



Voxengo GlissEQ **$69.95**
From the maker of SPAN comes an equalizer that not only has the same powerful spectral analysis capabilities as SPAN, but also boasts program-dependent band response. This dynamic approach to EQ makes GlissEQ ideal for boosting highs and lows.

FabFilter Pro-Q **$170**
One of the best all-round EQs on the market, Pro-Q is overflowing with features – up to 24 filter bands, zero-latency and linear phase modes, intelligent solo mode for auditioning single bands, and undo/redo and A/B comparison functions.

Waves Redd **$249**
Based on the EQ sections of Abbey Road Studios' mixing consoles, Redd delivers that classic '60s sound. As well as the simple EQ controls themselves, you also get Drive and Analog knobs for dialing in the perfect level of vintage warmth.

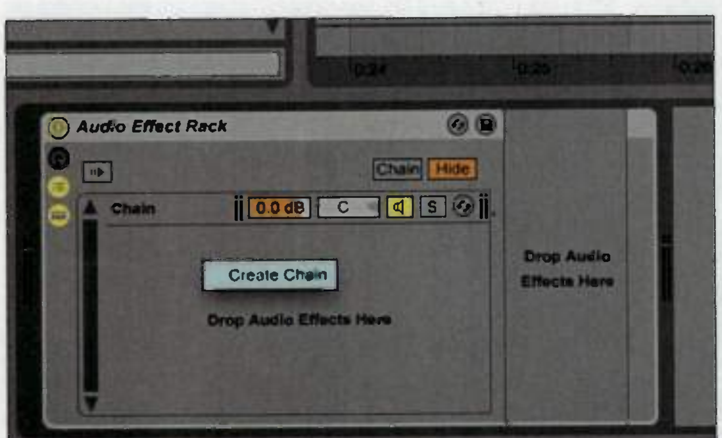
Step by step 4. Parallel processing in Live

1 Parallel processing is a technique where a processed (e.g. compressed) signal is mixed with the unprocessed version of itself. In the case of compression, this would give us a balance of loud 'wet' signal and punchy 'dry' signal. Start by dragging Funky jungle.wav onto an audio track in Ableton Live.

2 Drag Glue Compressor onto the track. The effect has a Dry/Wet knob, which means that we can mix some of the dry signal back in directly from the plugin itself. However, by setting up a parallel processing chain we can use multiple effects, which can then be mixed with the dry signal at the end.

3 Set the Glue Compressor's Threshold to -27dB, Makeup to 7dB, and Attack to 0.3. Add a Saturator after it, with its mode set to Sinoid Fold and its Drive turned up to 11.2dB. The processing we've added has boosted the peak level by a few dB, which we can check by bypassing the effect.

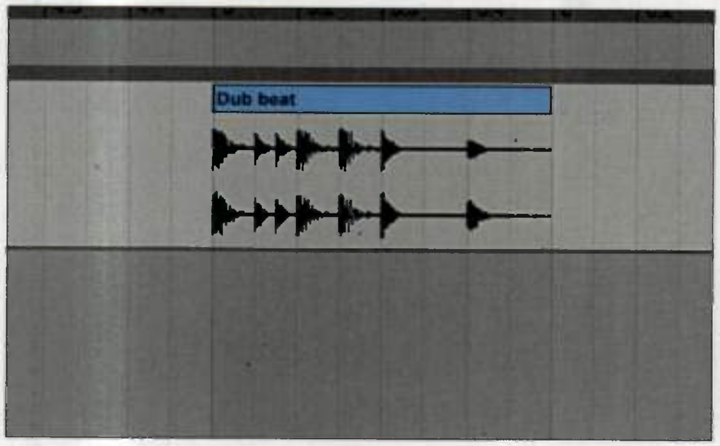
4 To compensate for this, turn the Saturator's Output down to -2dB. Now the signal peaks at around the same level as the unprocessed signal did, although the dynamic range has been significantly reduced, so it hits the peak level much more often. Now let's create our parallel routing. Select both effects, right-click and select Group.

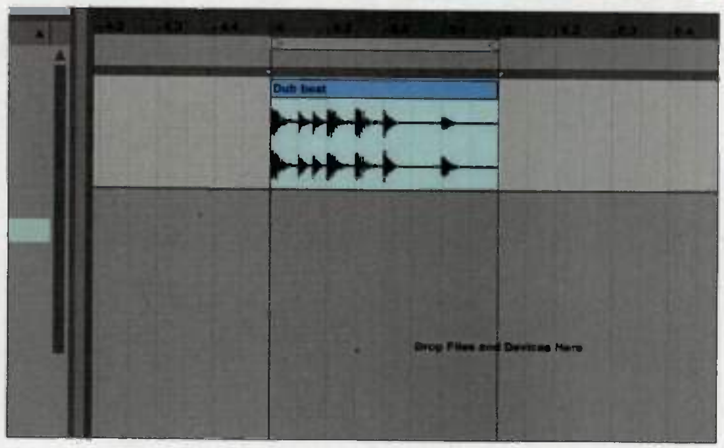


5 This puts both effects in an Audio Effects Rack. Click the Show/Hide Chain List button and a list of chains appears in the Audio Effects Rack. Currently, there's only one chain in the Rack. To add another, right-click below it and select Create Chain.

6 Now we've got our 'wet' chain with effects on it, and a new 'dry' chain for the unprocessed signal. As we've got two versions of the signal, the output of the track will be very loud, so turn both chains down to -6dB to halve their volume. You can use the Chain Volume parameters to balance the signals, and toggle the Audio Effects Rack's Device Activator button to compare the parallel processed signal with the unprocessed version.

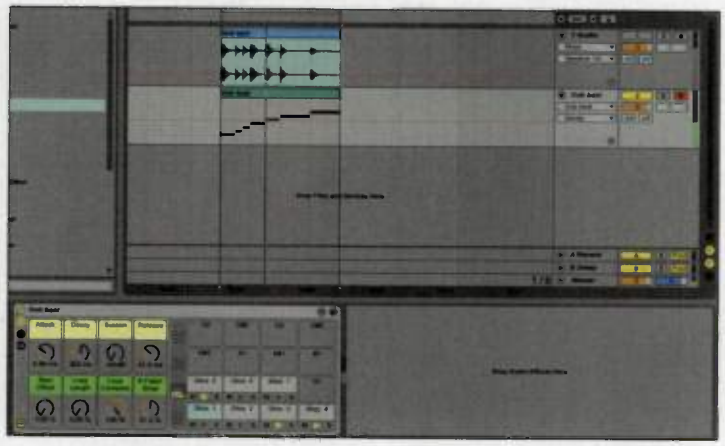
Step by step 5. Tightening drum loops with Live's Slice to MIDI function





1 Using drum loops or breaks is a convenient way to give your beats extra rhythm and character, but if they're too loose or have too much reverb, they'll need tightening up. Doing this manually can be fiddly, so some DAWs have their own beat-slicing capabilities to make it easier. Drag Dub beat.wav onto an audio track in Live.

2 Live will automatically set the project tempo to 140bpm. Select the clip and press Cmd/Ctrl+L to set loop points around it. This loop is drenched in dubby reverb, but we can quickly get rid of that and tweak the loop in various interesting ways by slicing it to MIDI.



3 Right-click the clip and select Slice to New MIDI Track. A window appears, asking how you want it to slice the loop and which preset to use for the resulting instrument. The default settings of slice by Transient and Built-in slicing preset will be fine for our purposes. Click OK to slice the loop.

4 A new MIDI track with an Instrument Rack loaded appears below the audio track. Mute the audio track. Now we can tighten up the new version of the beat on the MIDI track by turning its Sustain down to -infdB. Turn the Decay down to 362ms for a super-tight version of the beat.

MAKE YOUR OWN SLICING PRESETS

Live's built-in slicing preset is convenient because it enables quick access to the ADSR envelope controls of each slice, giving you an easy way to control a beat's dynamics. The problem with it, though, is that it doesn't have macro controls linked to each slice's Transpose and Detune parameters, meaning that if you want to retune the loop you have to go through and adjust every slice manually! Clearly, this isn't a very convenient solution, but thankfully it's possible to create your own slicing presets with which you can set up eight macros as you see fit. That's enough to control the ADSR and both tuning parameters, with two left over for other functions – e.g. volume and velocity sensitivity.

To make your own preset, create a new MIDI track and put a Drum Rack on it. Put a Simpler in the C1 slot, click the Show/Hide Macro Controls button, then the Map button on the device title bar to enter mapping mode. You can now assign macros by clicking a parameter, then clicking the Map button on a macro. Note that when you do this, the value on the parameter will reset to 0. When you've finished, exit map mode by clicking the Map button on the device title bar again, and set each macro to the default value you want it to have.

Finally, drag the Drum Rack into the User Library/Defaults/Slicing folder in Live's browser. This creates a preset, which will be highlighted, prompting you to enter a name for it. Type in a name, press Enter, and boom – you're done! Now, when you use Live's Slice to New MIDI Track function, you'll find the preset that you just created in the list of Slicing Presets. You can find all of our audio for these tutorials in the Tutorial Files at the link on this issue's contents page.

Step by step 6. Cutting or enhancing reverb with Logic's Enveloper

1 Apple's Logic Pro includes a unique and really quite powerful dynamics processing tool in the form of Enveloper. This can be used not only to tighten up the reverb on drum loops, but also to make them sound louder and boomier – ideal if you're after a big, 'live' drum sound. Start by dragging Dub beat.wav onto an audio track and setting the project tempo to 140bpm.

2 In the Inspector, click the Audio FX slot on the audio track and select Dynamics»Enveloper»Stereo. The plugin very effectively detects the transients in the loop, and enables you to control the tail of each hit using the Gain fader on the right-hand side of the interface. For starters, turn it down to -100%.

3 This immediately gets rid of the reverb and gives us a really tight, dead drum sound. You can get different curve shapes by using the Time knob to the left of the Gain fader. Set the Time to 60.00ms and the Gain to -74% for a less severe reverb reduction effect.

4 Let's use the effect to enhance the reverb now. Set the Time to 2000.00ms and turn the Gain up to 100%. As you can hear, this has a really drastic effect on the sound. The boominess is so loud that it's going to clip the master in fact, so set the Out Level to -9dB.

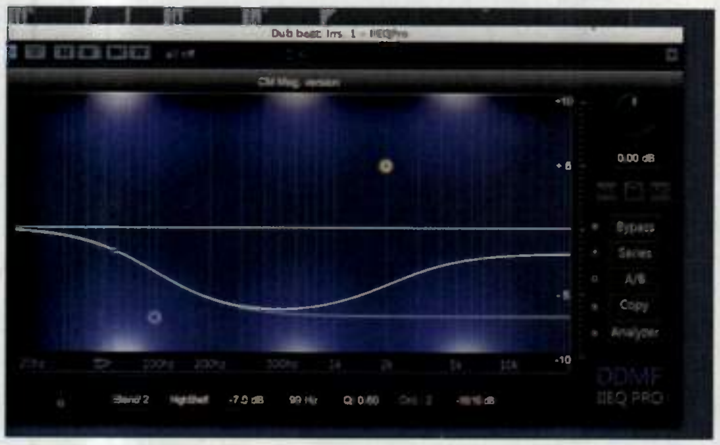
5 We can use Enveloper to treat the attack stage of each hit as well as the release stage. First, return the Gain fader on the right-hand side of the interface to 0%, then gradually turn the Gain fader on the left up as you play the beat back. Once you reach the 60% level you'll really start to hear each hit become more pronounced.

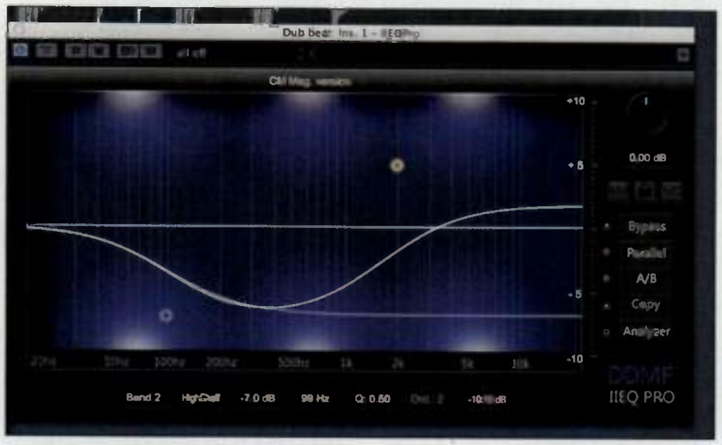
6 It's possible to dull the attack stage of the sound as well as enhance it. Turn the Gain fader down to 100% and it'll sound like you've turned up the attack time of each slice. This is potentially useful when you want to layer one-shot drums with a loop to give them character but the transients of the loops clash with those of the one-shots.

Step by step 7. High-shelving drum loops with parallel EQ

1 Using a pair of high-shelving EQs can be a very useful way to shape the high and low end of drum loops. To demonstrate this, we're going to use DDMF's IIEQPro CM – available free from our sister magazine Computer Music – but any similar EQ will do the job. Create a new audio track in your DAW and drag Dub beat.wav onto it.

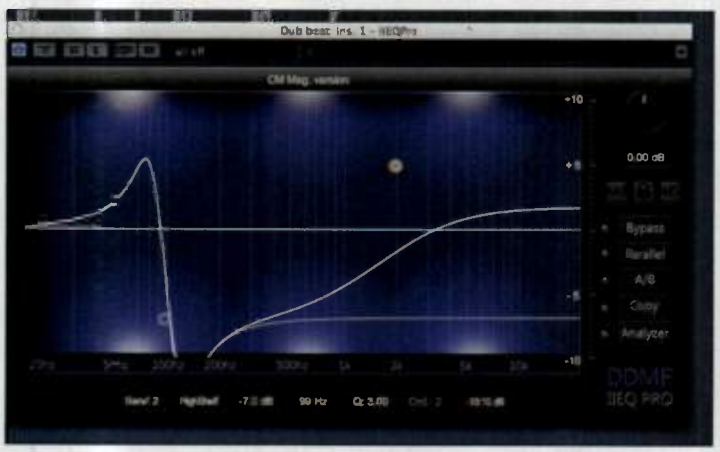
2 Insert the EQ onto the audio track. We select a high shelf EQ and an orange node appears at the left-hand side of the graphical display above. Drag this to the +5dB position at 2000Hz.

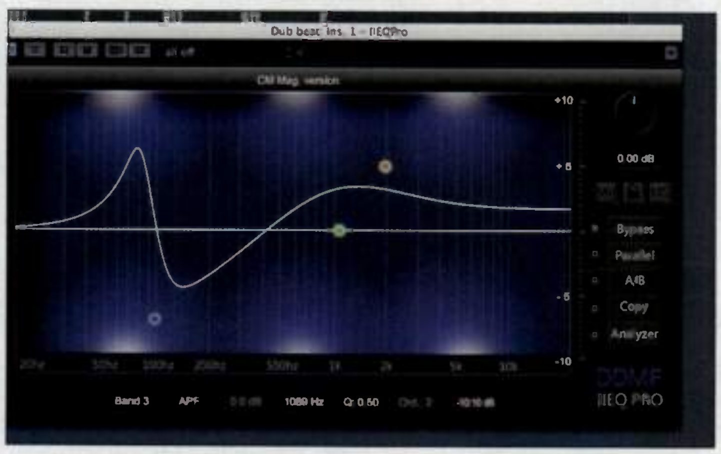




3 Now create a second high shelf band and this time move the node to -7dB at 300Hz. The white line displays the overall effect of the two bands, revealing that by using two high-shelf EQs we've created a curve that we couldn't have made using just one.

4 IIEQPro CM has an unusual Parallel mode. In this mode, each band works on a dry version of the signal rather than one that's already been processed. This affects the overall sound of the EQ. Create a similar effect in any DAW using two parallel effect channels.





5 As well as using the Frequency and Gain parameters, we can also use Q to control the shape of the EQ. For an extreme example, by tuning Band 2's Q up to 3.00 we can remove the low end of the reverb from the beat while simultaneously boosting the kick.

6 Finally, let's try out an all-pass filter mode. In serial mode this doesn't do a whole lot, but in parallel mode it has a quite dramatic effect. Select Band 3 and set it to APF. Now, when you sweep its frequency up and down, you'll see that the overall EQ warps around its position – this is ideal for experimenting with different frequency responses.

Eris Drew

By Danny Turner

One of the few DJs still performing with turntables, house producer Eris Drew also uses her Technics as a production tool. Danny Turner delves into her debut album

In common with many DJs, the pandemic obliterated Eris Drew's schedule, which had recently expanded to almost a hundred gigs globally. So Drew took the bold decision to trade her hometown of Chicago for a secluded forest cabin in New Hampshire, living with production and label partner Octo Octa (Maya Bouldry-Morrison), and started work on her debut album. Titled *Quivering In Time*, Drew's solo effort feels not too dissimilar to her DJ sets, fast-moving percussion-heavy dancefloor tunes alight with upbeat party vibes. With her tracks built from a stack of vinyl samples, Drew added kicks, scratches, vocal samples, hand-played keyboard riffs and percussion to recreate the communal euphoria of a club scene slowly returning to life.

Tell us about the move from Chicago to New Hampshire on the east coast...

The practical story is that I fell in love and wanted to move here, but there's a spiritual story too, as for years I dreamt of having a life where I could be in nature and make music but still have some connection to cities and DJing. As a Chicago girl, the move seemed so irrational to me because the music I make is so in the bones of the city and leaving seemed something distant for my older years, but I fell in love with Maya who grew up here and ended up in rural New Hampshire in a log cabin. I look out my window now and it's just forest.

Your partner Maya published a DIY guide to building a home studio. Has that helped you both settle into this new environment?

I wouldn't say the guide was related to that, but it's the first time I found an environment where I shared a studio in a space with someone else, so there was constant collaboration and discussion about music. The cabin itself really does affect our music-making experience because it's kind of like setting up shop inside the body of a guitar. It's an extremely resonant space. My subwoofer is on an all-timber floor and I've got slats above me that bounce the sound around. It's an extremely warm-sounding environment that colors the sound, but as a songwriter I love it. We're not so remote that we don't have any home comforts and there's a generator in the basement that we only need to run in emergencies.

Presumably, at odds with your previously hectic Chicago lifestyle?

I'd been in Chicago for so long and it's a tough town but I'm so grateful for my experiences. Having lived in the city for 25 years you get to kind of know everybody, but the club scene is tricky there now because it's shrunk quite a bit. There are only a few venues and party crews that support this kind of music, so even though Chicago is huge, it feels like a small city because a lot of people want to work but aren't able to.

> "There's been less focus globally on what people were doing in Chicago and more of the big commercial clubs got to be popular."

In Chicago house music has receded in popularity and certain venues are closing...

I could probably write a book on it. One chapter would be about gentrification, the changing of the neighborhoods and pushing out of venues, but there's also been a change in the music scene. I'd always hear the house-heads saying that by the late '90s the music had really changed and I can't say that's wrong. There's been less focus globally on what people were doing in Chicago and more of the big commercial clubs got to be popular. The whole bottle service thing was huge in the late '90s and '00s and that's been very pervasive too. If it wasn't for Smartbar and some of these other places, there wouldn't be anywhere to have that true, elevated house music experience. Another factor is the legal environment because they passed the rave ordinance in the late '90s, which basically made it really easy for police to stop parties. Not only could they fine a venue or promoter, but they could fine the DJ and that chilled the hell out of everyone because they could potentially be subjected to a $10,000 penalty. People still do it, but you either have to keep things really small or have connections.

Your label's called T4T LUV NRG. Were you a fan of the original Hi-NRG sound?

I love Hi-NRG music and the sound that always pops into my head is NRG's He Never Lost His Hardcore, which was very rave. Being in Chicago, the proto-house sound was very much based on European imports like Euro disco, Italo, NY electro and older disco records. Patrick Cowley's Hi-NRG sound was also very much in the air. My mum loved all that music – she was a huge Sylvester fan, and not just the big songs but all the 12 mixes. We used to jam out to Take Me To Heaven in the car.

So your parents were the source of your love of disco and dance music?

I was a little kid living in Minnesota in the '70s and my parents liked to go disco dancing at night. They'd get a babysitter, go out early and, as far as I can tell, found a pretty good club because they had a neat, idiosyncratic collection. The first disco record I remember hearing was C.J. &

Company's Devil's Gun, which is a pretty dope underground record. My parents were straight, white people, but from a young age my mum told me, 'If you wanna really hear good music Eris, you gotta go places where there are black folks and gay folks'. That was a good thing to hear in the early '80s because I was ridiculed for my music tastes as a kid. It was cool to like Guns N' Roses and shit like that, but there was I with my Depeche Mode shirt listening to house music mixes on the radio late at night.

What was your general mindset when it came to creating what is your debut album?

I'd been thinking a lot about how I wanted it to be structurally. I love albums so much and felt like I wanted to do a dance album that was different to all the others. When 12" artists go to make an album it's so often about having guest appearances, collaborations or songs they wouldn't necessarily play on a dancefloor. I wanted an album that was programmed, hyper-arranged and I guess it's clichéd to say it but a journey from front to back. A few examples would be Orbital's *The Brown Album*, 808 State's *ex:el* or Run DMC's *Tougher Than Leather*, which all have quite a bit of evolution within them.

It definitely has a celebratory tone. Was that a reaction to the global circumstances?

I started writing a couple of songs on tour – just little drum samples and working on chords and melodies. When we went into quarantine I had one track called 'Quivering At The End Of Time', which was a reference to the psychedelic notion that maybe a historical period was ending. I had no idea that I was about to head into lockdown, but I thought it was a prescient moment. I didn't want to just write an album that's dark; music's not my way of expressing the dark side of my psyche; it's more about transforming it.

Trying to turn a negative into a positive?
Well, it's not like I've been out here in the woods all happy over the past 18 months – our music careers suddenly came to a slamming halt. Having toured the world in 30 countries, Maya and I were on a real high. We're both transgender women who'd felt a whole lot of triumph and momentum, so it felt like we were telling a freight train to stop. From all of that frustration and feelings of loss – and even though I was feeling quite depressed, I wanted to take all of the positive energy of the last year and a half and cast it back into the music.

Are you now able to view your enforced break from live shows as an inadvertent positive?
All I know is that I took the time to record an album, something I never thought I'd have time to do. Also, I was not what I would consider an engineer. I would mix down my tracks as best I could, but on the previous *Fluids of Emotion* EP I got the opportunity to work with an amazing sound engineer in Detroit called BMG. We spent two weeks working together, not just to engineer the record but so I could use the time to create my own engineering practice. It was very hands-on and we were both behind the controls, so I really learned a lot about things like compression and balance.

The production is very accomplished so you appear to have learned a hell of a lot in a relatively short time.
I wanted to make a very hefty and bassy record but one that still had a lot of clarity and dynamics, so it gave me the chance to do all the trial and error I needed to get my ears and skills in shape. But I have to say that Maya and I must have listened to these songs a thousand times in the car and talked about them, so she's kind of the 'executive producer'. I would work on the songs, bounce them out and listen to them in the car, on headphones and the laptop. They call it A/B-ing, but I was A/B/C/D-ing and did that for a solid year and a half.

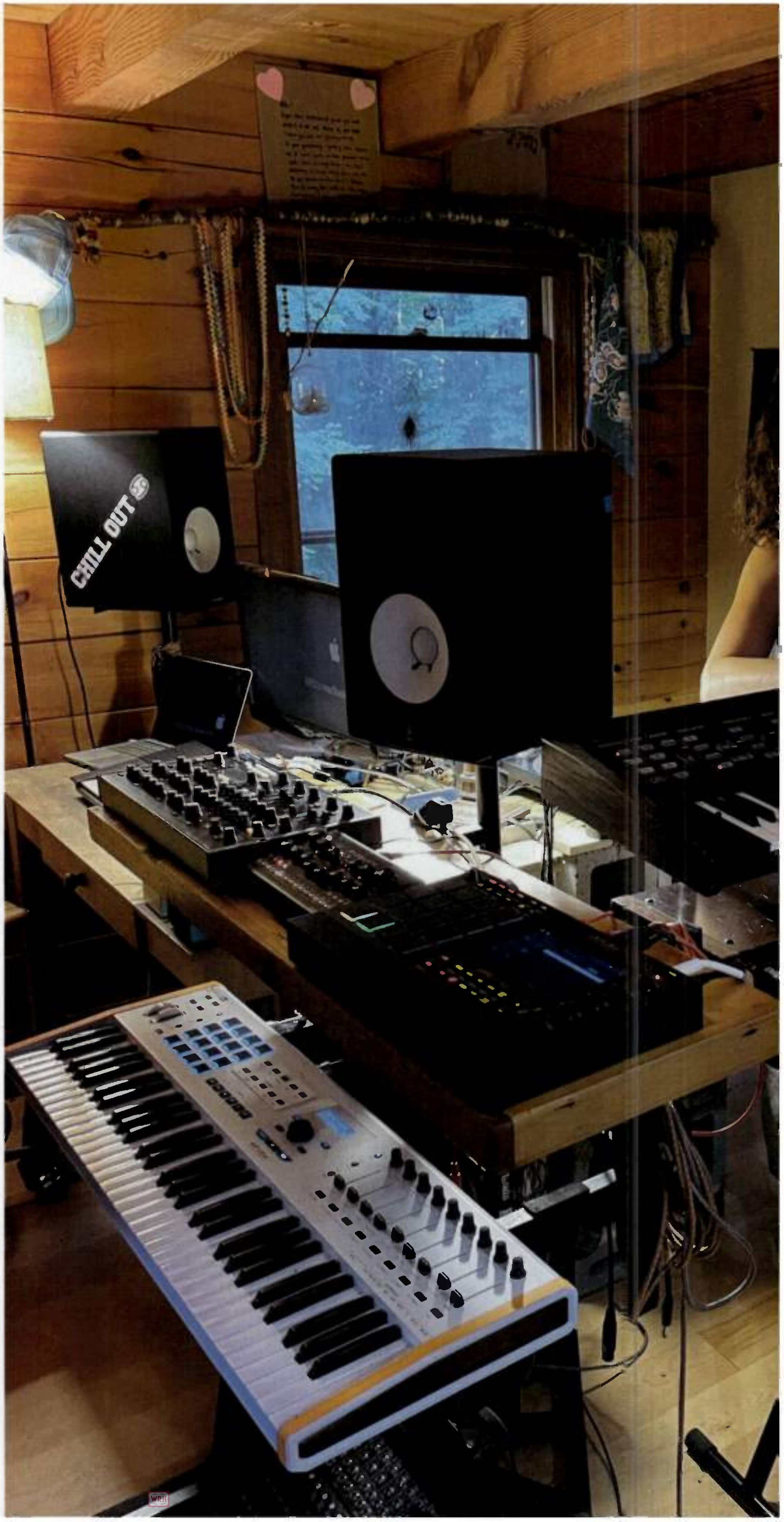
The album almost replicates a DJ set. The energy is relentless, yet each track is independent of another. Did you think about blending it all together?
Each track should flow into one another but I hadn't thought of making it fully continuous. I'm very much a DJ at heart and like to be able to look at the vinyl, see the clean start and play the track from the first beat. I wrote 12 songs and knocked them down to 9, and part of that was based on how we thought the songs flowed and not wanting to repeat certain themes. It also feels like DJing because of the way I make music. I use samplers but not necessarily in the way everybody else does because I use turntables instead of timestretching. The average vinyl DJ is working in a 0- 8% time change range – and there are a few Technics decks where you can switch between 8-15% – but mine are (+/-) 50% and I really take advantage of that.

How do you use turntables to make a track?
I start with a simple beat collage, which comprises of a few different rhythms that I'm mixing together. I'll lay one down and mix over it while doing some multi-tracking in the computer. Then I get out my keyboards and start to write, so the songs are still individualized because I have a songwriter's approach. I don't use step sequencers, everything is sequenced in real-time and I'll often work on chords on the piano. If I had to put it simply it's keyboard jams to my DJing with added post-production.

So you're using turntables to play the initial beats and multi-layering them in the DAW?
The first part is pretty simple. I'll find a little beat or a part of a beat and pick a song tempo, so instead of randomly sampling beats into the computer and changing the pitch to match whatever I decided the tempo is going to be, I'll make that decision up front. That first beat goes in and I might cut that up and repeat it over a few measures, then I'll grab another record and do what I'd normally do in a set, which might be to drop in a snare at a weird place. For example, the track Baby has all these snare drops at the start but I didn't swing those, the legato comes from me throwing them in a little bit late by scratching and dropping snare drums from different tracks. One of my techniques is to adjust the record's speed, so I might record the snare at its normal pitch or really pitch it down, up or at half-time, so there's tons of half-time and double-time breaks all over the record and they're created on-the-fly rather than detuned in the computer.

Do you prefer to create quite a rough mix with all the inaccuracies that may come from sampling included?
There is plenty of editing that happens in the computer afterwards, but it's really just editing it so the sounds are where I intended them to be. For example, if I scratched a scratch a little late then I'll just move it to where I wanted it to be. But the recordings are the recordings, so they have noise in them and timing issues because I used a lot of live drum samples, but I don't get fussy about that. The records I loved have tons of samples doing their own thing and it's that funk and energy that gives them a feel that's so different to a lot of modern music.

Many of the tracks are notable for the clever use of vocal samples — where are you sourcing them from?
On the title track, I used samples from a series called *The Ultimate Skratch Record*, which were these battle records that were very popular with house DJs. That's kind of unusual as we tend to think of battle records coming from hip-hop, but some were made in Chicago that had lots of beats and samples from house songs but also classic hip-hop samples. For example, there's a sample on 'Quivering In Time' that sounds like a girl saying 'bass', but it's actually Chuck D sped up. I don't like to use typical sounds. That track starts with quite dry Florida breaks plus a 'hah' hip-hop beat, but the next sound you hear, which sounds like a shaker, is actually the tremolated sound of a meadow, and one track has a hippo



growling at one point amongst a bunch of Run DMC drops – so there's all kinds of stuff in there.

Will you add in percussive elements using drum machines or software-based tools?
Yes, I have a few rompler-style drum machines that I love and have used for years. I used a Roland R-8 MKII from the '80s, which is arguably their last great drum machine. It has wonderfully clean 808 and 909 sounds and they're very particular. I also used a drum machine called the Yamaha RM50 and played some drums using my Nord Drum 2, which I love because, unlike the romplers, it has a synth brain and you can synthesize sounds from the ground up like you would with a Moog. I do use some softsynths too and really like D16's Drumazon and Nepheton, which are 808 and 909 clones. They're great units for when I need a clean sound really fast. I like it when one second a sound is dirty but the next sound has really clean hats or kick drums. That's why I love Run DMC's Tougher than Leather album, because one second you have these really great rock 'n' roll samples and then a really clean 808 comes in over them.

Are you still a vinyl collector or do you feel there's not enough new vinyl being produced that's worth sampling from these days?
We went to see my parents and brought my collection back with me, which we think is about 6,000 records. We have off-site storage and one of our projects over quarantine was to go through my entire collection, organise it by year and start putting it into Discogs. Most of my samples are from older records; I can't think of a single sample that I used from a record made by a producer that's working today. I'm a 100% vinyl player on tour and there's very few of us left. I've been doing this so long and have developed such a skill when it comes to mixing with turntables that I don't want to let it go. My mum was an antique dealer and told me not to believe that every time something's considered as progress it actually is. It's worth mentioning that I sample off CDs too and keep an old DVD player in my studio hooked up to my mixer.

When did you first get into keyboards?
The earliest experience I remember was in my grandmother's basement because my grandfather had one of those old player pianos. You put a scroll in and could make it sound so demented by controlling the pedal. My grandmother also got me a little Casio toy when I was in third grade and I loved this little thing. Once I had a little of my own money I went to a store called Service Merchandise that had the Yamaha PortaSound synth series and I've still got the PSS-480 in front of me now. It's a two-operator FM synth with nine controls that you can use to change the sound, store five sounds and it's got a five-track sequencer on it, which was a lot for $100. That's what I started to teach myself synthesis on – stuff like Depeche Mode, Information Society and industrial music. I wanted to be in a band that made that kind of music, but I love the keyboard to this day. It's a noisy little bugger, but on the song 'A Howling Wind' every single synth sound is from it because it has great singular tones. My hallmark synthesizer is the Chroma Polaris...

The mid-'80s analog Fender/Rhodes synth?
Yes and there's a neat history there. ARP was basically going out of business and they came up with this polysynth that they'd already designed. It's an analog synth with an ARP sound and a digital stepping filter. It's the opposite of how everyone wanted to structure a synth at the time as it has really wonderful digital oscillators with an analogue filter, which gives it such a wild and different sound because when you filter it has to do these little digital '80s calculations. The opposite of that would be my Yamaha rack unit – the TX816, which is basically eight DX7s in a box. It gives you the cleanest tones you can imagine for when you want something that sounds like it's coming out of a video game.

What DAW/plugins are you using these days?
Logic 10, because I just need a glorified tape recorder that can run plugins in time code. I'm more of a player than a programmer, so I'll draw some dynamics automation and programme hi-hats, but it's mostly volume changes and some wet/dry effects. I use some pedals for processing. I've got a Lexicon rack, which I haven't used in a while, but I keep a little pedal array and when it's time to work on a keyboard part I might run them through one of those. My favorites are a little tremolo plugin and a delay plug called an Aqua-Puss.

Do you see your immediate future as a producer rather than DJ?
I've been touring for most of the past nine weeks now. The second we get off the phone I'm jumping in a car to go play a three-day rave in upstate New York, so we're back at it and my schedule's absolutely full until the end of the year. Production-wise, if I'm moving around a lot I'll create collages, put them together on the road and engineer them when I get back... With varying success! ■

THE ART OF SYNTH SOLOING

An OG Crossover Master

A musical toast to the synth innovations of Jeff Lorber...

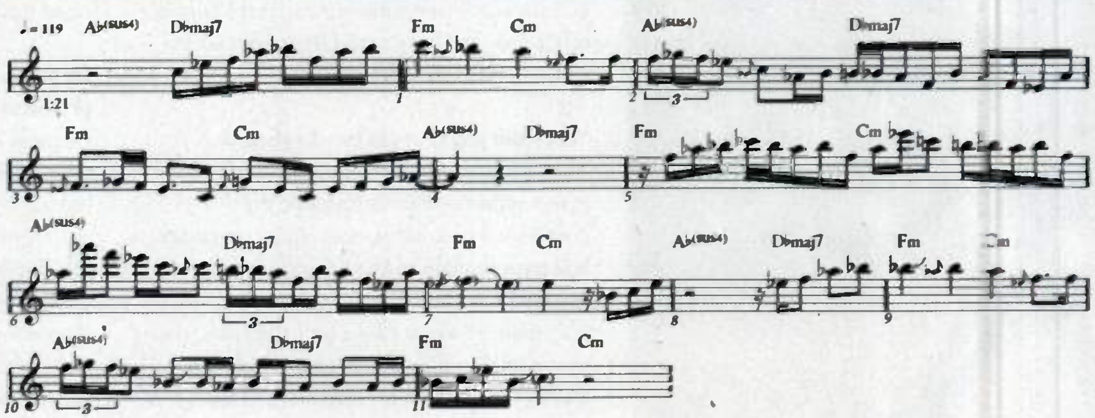
By Jerry Kovarsky

A number of times over the last 50+ years, an artist has come along who absorbed the vocabulary and energy of the music around them, and brought it greater commercial success as instrumental music. I am thinking of artists like Horace Silver, Ramsey Lewis, and Booker T. and the MG's. To that list I add Jeff Lorber. Jeff took the energy and rhythmic foundation of funk, adding to it his formidable jazz chops, and wrote highly melodic tunes to feature it all. While he has certainly made some commercial music over the years, Jeff has returned to his roots over the last two decades, making sophisticated and fiery music once again under the name The Jeff Lorber Fusion. Let's explore some of my favorite Lorber licks from his defining early years.

Equal Parts Funk and Jazz

From his emergence in 1977 through the mid-80's Jeff played synths on his recordings, often soloing with them. He was never a pitch-bend jockey: he always employed it sparingly, laying down tasty, melodic solo lines. A large part of that style was based on the minor Pentatonic, and

Example 1 Jeff's solo on "Always There", from the It's A Fact release.



Blues scales, and he developed his lines wonderfully. Take a look at **Example 1**, the opening of his solo on "Always There", a classic Ronnie Laws composition Jeff covered on the *It's A Fact* album in 1982. He opens with an F minor Pentatonic riff starting as a pick-up into the progression, which slips into one of his signature bluesy licks in the second part of bar 2, and continues through bar 3. Notice his use of grace notes as well. He leaves some space and then builds his next phrase from part of the previous opening figure, giving the solo structure and a theme that he's playing off of. And he keeps working that same bluesy figure. His next phrase (starting in bar 8) again plays off the same theme as his opening line, but here he goes back to starting his phrase as a pick-up, giving his phrasing nice variety. While it is not overly complex playing, it is highly melodic with well-paced and structured development.

Leaning Into His Jazz Roots

Without a doubt, my favorite solo is on his tune "The Samba" from *Soft Space* (1978), the second release by the Jeff Lorber Fusion. Interestingly, when I told Jeff I was covering this solo he told me that he pre-composed it, wanting to have a "perfect take" that was well-thought-out.

Example 2 shows the opening of his solo, which works the F# minor Pentatonic and Blues scales over the more jazzy chord changes. In bar 4 he moves a bit jazzier, outlining the D Major seventh nicely and then moving into a more bebop-based line for the C# Dominant with the altered color tones. That line is based on the C# Altered Dominant scale, which comes from the jazz vocabulary (see **Example 3**), and is a favorite of Jeff's. It resolves very colorfully into the ninth of the next chord. In bar 5 he returns to bluesier soloing, only to come back to the jazz in bar 6, with a tasty ii-V-i lick which comes straight from the bebop language.

Notice how he uses he same Altered Dominant scale, but this time it's over the G7, which in jazz pedagogy is called a tri-tone substitution for the C#7 chord. The line resolves into the F# minor using the major seventh tone, a wonderful jazz color. Not to lose the listener, he finishes up with some straight Pentatonic; such a perfect blend of the two worlds.

Example 2 The opening section of Lorber's solo on "The Samba", showing his masterful blend of blues/funk and classic jazz vocabulary.

Example 3 The Altered Dominant jazz scale, which is used over a Dominant seventh chord with altered ninths and thirteenths.

Example 4 Billy's first solo

> "Without a doubt, my favorite solo is on his tune "The Samba" from *Soft Space* (1978), the second release by the Jeff Lorber Fusion"

Playing The Key Center

Example 4 is taken from "The Magician", also from his *It's A Fact* recording. It gives us a chance to see how Pentatonic/Blues lines work on a tune that is in a major tonality. And Jeff is not trying to play the changes; he is soloing over the key center. The first four bars are pretty straightforward — it doesn't matter if you think of the scale as a Bb Major Pentatonic or G minor Pentatonic, it's the same notes. This is an example of superimposing a different Pentatonic scale over a key center (Bb/Gm over the key of Eb), which adds some nice color tones (the 6th and major 7th). The grace notes and slight bends add some soul to these great lines. Leading into bar 5 Jeff brings the jazz, with a 2-bar phrase that superimposes some classic be-bop for a Cm7 to F7 altered sound, which is the ii-V7 in the key of Bb. Moving into bar 7, he plays a tasty little Bb blues lick to finish the phrase out. ■

EASY GUIDE

Consonance and dissonance

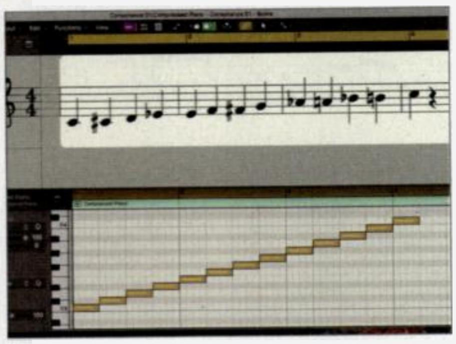
Our incumbent theory professor examines how a combination of nice and nasty can provide the ultimate musical contrast

Like most art forms, music is all about light and shade – you need a blend of both. In this regard, two of the main things to consider when composing a piece of music are consonance and dissonance. It's all to do with the intervals between note pitches. If we take two notes and count the number of semitones between their pitches, the result is known as an interval, and each different interval can be labeled either consonant or dissonant.

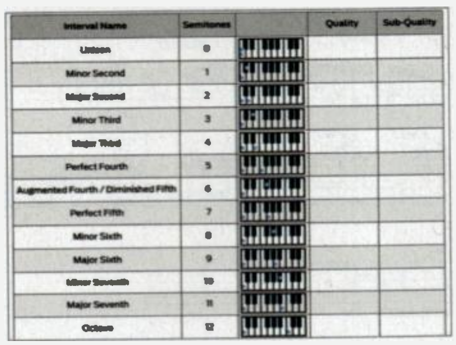
The dictionary definition of consonance is 'harmony and agreement among components', so consonant intervals are usually pleasant to listen to, fostering a sense of agreeable wellbeing and satisfaction. A dissonant interval, however, just generally makes you wince, and the tension it creates just makes you want to hear a consonant interval as soon as possible after it – a concept that's known in music theory circles as resolution. Dissonance can be used to great effect in composition to evoke emotion. Pretty much the whole spectrum – from wistful sadness and poignancy through to outright fear and terror (Psycho shower scene anyone?) – can be transmitted through artful use of it.

This month, then, I'll delve into which intervals qualify as consonant or dissonant, and show a couple of ways in which you can exploit the contrast between the two, to add a touch of sophistication to the music you create.

Step by step Exploring consonance and dissonance

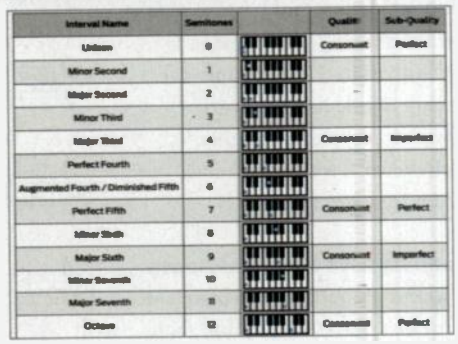


01 Rather than kick off this month with our usual C major scale, this time round we're starting with a chromatic scale – all the notes, including the black ones, from C to C on the piano keyboard, totalling 12 in all (well, 13 if you count the high octave of C).



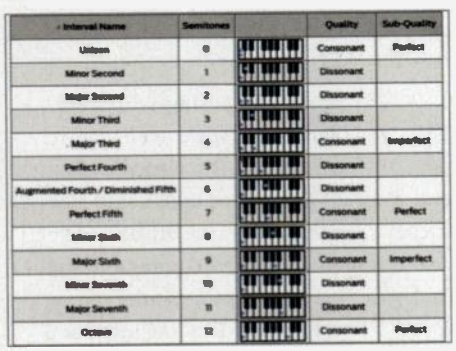
Interval Name	Semitones		Quality	Sub-Quality
Unison	0			
Minor Second	1			
Major Second	2			
Minor Third	3			
Major Third	4			
Perfect Fourth	5			
Augmented Fourth / Diminished Fifth	6			
Perfect Fifth	7			
Minor Sixth	8			
Major Sixth	9			
Minor Seventh	10			
Major Seventh	11			
Octave	12			

02 I've already discussed intervals at some length in previous Easy Guides, so here's a quick refresher. Every note in the chromatic scale is a set interval, or number of semitones, above the root note. This chart shows the names of all of the intervals that occur in an octave.



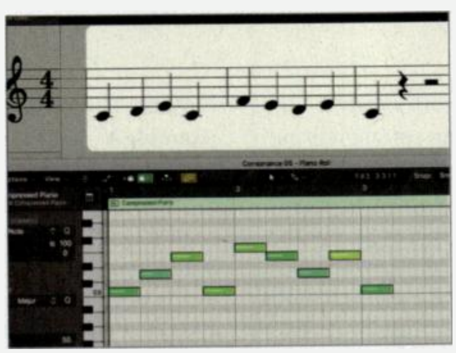
Interval Name	Semitones		Quality	Sub-Quality
Unison	0		Consonant	Perfect
Minor Second	1			
Major Second	2			
Minor Third	3			
Major Third	4		Consonant	Imperfect
Perfect Fourth	5			
Augmented Fourth / Diminished Fifth	6			
Perfect Fifth	7		Consonant	Perfect
Minor Sixth	8			
Major Sixth	9		Consonant	Imperfect
Minor Seventh	10			
Major Seventh	11			
Octave	12		Consonant	Perfect

03 Intervals come in two categories – consonant and dissonant. The consonant (ie, pleasant-sounding) intervals can be further separated as perfect and imperfect. Perfect consonances include the perfect fifth and octave, while the major third and sixth are in the imperfect camp.

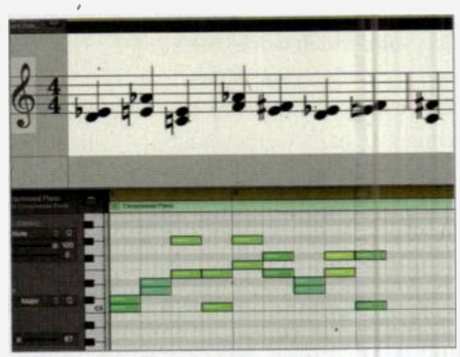


Interval Name	Semitones		Quality	Sub-Quality
Unison	0		Consonant	Perfect
Minor Second	1		Dissonant	
Major Second	2		Dissonant	
Minor Third	3		Dissonant	
Major Third	4		Consonant	Imperfect
Perfect Fourth	5		Dissonant	
Augmented Fourth / Diminished Fifth	6		Dissonant	
Perfect Fifth	7		Consonant	Perfect
Minor Sixth	8		Dissonant	
Major Sixth	9		Consonant	Imperfect
Minor Seventh	10		Dissonant	
Major Seventh	11		Dissonant	
Octave	12		Consonant	Perfect

04 Dissonant intervals include the minor second (a single semitone), major second (two semitones) and major and minor sevenths. Let's not forget the evil-sounding tritone, too, also known as the augmented fourth or diminished fifth – an interval of six semitones, or exactly half the octave. Played out of context, these intervals sound jarring and unsettling.



05 We can put this theory to good use. For a second, let's compare dissonance to, say, chilli powder. Eat it by itself, and it's not a very pleasant experience. Add a dash of it here and there to other dishes, though, and it can make all the difference. To illustrate, here's a short, single-note melody containing notes taken from the C major scale.



06 I'll harmonize the notes in two-part harmony, but using only dissonant intervals. It now sounds like something out of your worst nightmares. This is an extreme example, though – remember the chilli analogy? Dissonance can be thought of as musical seasoning – if we didn't use it at all, music would be extremely bland.

Recommended listening

Avicii - WAITING FOR LOVE

Subtle chord variation here: F#m - C#add11 - D, the dissonant F-F# interval in the second chord emphasizes the consonant D major that follows.

bit.ly/avicii_waiting

Rihanna - DISTURBIA

For a not-so-subtle illustration of consonance/dissonance, check out the deliberately terrifying dissonant intro to this tune. As soon as the beat kicks in, consonance rules.

bit.ly/rihanna_dist

Pro tips

Power up

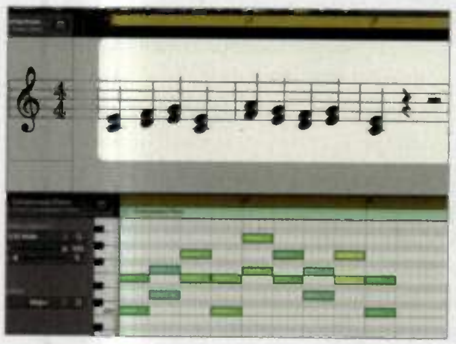
Generally, the more consonant the interval, the more it sounds to the average listener like a single tone, rather than two. Take guitar power chords, for example, which are made up of the root and fifth of the chord, with no third present. Since the perfect fifth is a clear contender for the most consonant interval in all the land, power chords sound like just one note to the untrained ear. A minor second, however, probably the most dissonant interval, is very definitely perceivable as two notes.

Inversion therapy

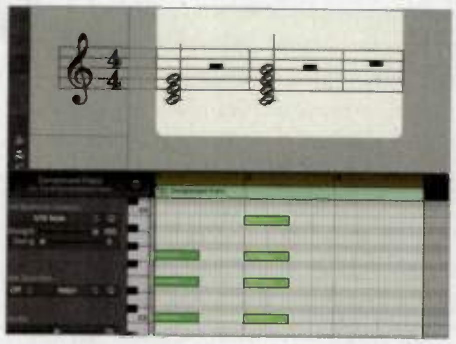
The minor ninth (the interval one semitone higher than the octave) is known as an 'avoid' note – it's considered just too dissonant. This can be got around by inverting the interval and playing a major seventh instead. So, the interval between the E and F in a Cadd11 chord is a minor ninth, but playing the F below the E, not above it, turns the interval into a more palatable major seventh.

By Dave Clews

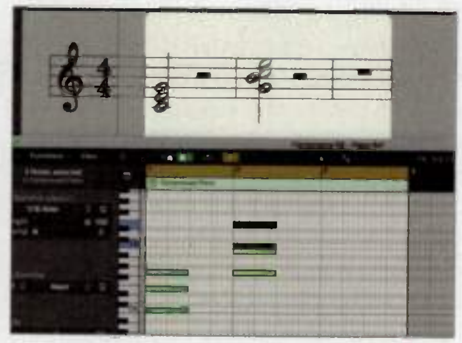
Over the course of his 25-year career, Dave has engineered, programed and played keyboards for numerous artists including George Michael and Tina Turner



07 Here's the same tune harmonized with more consonant intervals. It's much more cheerful, due to its major key and consonant intervals between melody and harmony. But, note the second and third notes, D and E have been harmonized with minor thirds – dissonant intervals. They work here, as they use notes diatonic to C major, the key we're in.



08 One of the most common everyday ways we use dissonance is when voicing extended chords. Take a major seventh chord, for example. The regular voicing of a major triad goes root, major third, fifth – all consonant intervals. Adding a dissonant major seventh shouldn't work, but it makes for a considerasbly more grown-up chord in the shape of the major seventh.



09 The example you can see above is Cmaj7 – C, E, G, B. If we were to invert the chord to the second inversion, by moving the lowest two tones up one octave to get G, B, C, E, we would get a cluster of two notes in the middle of the chord – B and C – that are effectively separated by an even more dissonant minor second interval.

10 Here's a short progression in the key of A minor – the relative minor of C major. We've got the chords F major - E minor - E minor - F major. Currently, the chords are voiced as regular triads, containing just the root, third and fifth of each chord. So, with the exception of the odd minor third, the intervals are therefore mostly consonant.

11 To spice things up, I'll add a major seventh melody note (E) to the top of the first F major chord, making Fmaj7. I slip in a passing Dm7/G chord to underpin the melody, before adding a C to the first Em chord and raising the bass note from E to A to make an Am9. The dissonance here is the minor second interval between the B and C notes.

12 In the second of the Em chords, I use the melody note of D to effectively move the consonant fifth (B) to a dissonant minor seventh (D), destabilizing the chord a little and giving it the character of an Em7. Meanwhile, the final F major chord receives a G note between the F root and the melody note of A, making an F9 chord.

BLAST FROM THE PAST

Fairlight CMI

A trio of intrepid Australians set out to create a digital synthesizer but ended up with an entirely different instrument – one that changed music forever

TECH SPECS

Year of manufacture
1979-19[illegible]

Original sale value
$25,000+

Current price
New, $20,000;
Original, $6,000+

It's impossible to overstate the enormous impact of sampling, and hard to imagine a world without it. At the start of it all, there was the Fairlight CMI. The Fairlight wasn't the first device capable of recording and playing back a digital sample (Publison's popular DHM 89 B2 sampling delay processor hit the streets in 1978, with the Fairlight following in 1979), but it was the first to offer a polyphonic, keyboard-based interface.

In what has become something of a legendary tale, sampling was, in fact, only added to the Fairlight's repertoire at the last moment. Initially, it was intended as an eight-voice digital synth with customizable waveforms that could be drawn on a CRT screen with a futuristic-looking light-pen. Designed by Peter Vogel and Kim Ryrie, the Fairlight used digital technology licensed from Creative Strategies' Tony Furse, who'd spent years designing a digital synthesizer for the Canberra School of Music. Furse's synth was the Qasar M8, a massive, hand-wired black box measuring four feet across and a foot-and-a-half in height, and stuffed with 20 8" x 8" circuit boards plus a fan to keep its 64KB of RAM cool.

Though Fairlight was founded in 1975, it would take years for Ryrie and Vogel to get their "Computer Musical Instrument" to market. They'd already taken pre-orders before they realized that a last minute circuit re-jig would allow for rudimentary sampling. This in turn prompted a complete overhaul of the whole thing, this time focusing on the instrument's new 8-bit sampling capability.

The now-iconic multicomponent Fairlight CMI system finally saw the light of day at the end of 1979, just in time for the technology-obsessed '80s. It was the perfect match for the decade to come, with its sleek ivory cabinetry, rakishly-angled CRT terminal and the ever-present light-pen.

However, such bleeding-edge tech came at a cost. Fairlight systems started at $25,000 with some configurations costing as much as $175k! Needless to say, only hitmakers could afford the

things, ensuring that the Fairlight's distinctive sound would be imprinted on the popular music of the day. Users such as Peter Gabriel, Kate Bush, Frankie Goes to Hollywood, Jan Hammer, The Cars, David Bowie and Herbie Hancock exploited the fresh instrument's then-novel sampled sounds as well as its real-time 'Page R' sequencer.

The rest of us could only dream of sampling until the price came down, which it finally did thanks to other manufacturers, eager to capitalize on the Fairlight's Brave New Sound.

Most people carry more powerful computers in their pockets than the boys at Fairlight could ever have dreamed of – yet as we gaze out over our desktop DAW with its attached QWERTY keyboard and MIDI controller, we can't help but chuckle at its vague resemblance to the sampler that started it all. ■

Three great plugin alternatives

Peter Vogel Instruments CMI **$50** Everything old is new again and you can, as it happens, buy a new Fairlight CMI from Peter Vogel's company for the princely sum of just under $20,000. If you don't have that lying around, you can grab this virtual version for iOS. It features a recreation of the Page R sequencer and the entire Fairlight CMI IIX sound library (over 500 "voices"). **petervogelinstruments.com**

UVI Darklight IIX **$199** UVI has recreated the Fairlight as a software sampler for the modern desktop DAW, providing a whopping 2.34 GB library of samples that can be played through one of three virtual instruments bearing the familiar Fairlight color scheme. Synthesis and drum/phrase sequencing are included, as are modeled filters, LFOs, envelopes and effects. **www.uvi.net**

Sonic Bloom CMI Live Packs **Free** These ten Fairlight sample packs are available free from musician, singer, composer and multi-instrumentalist Madeleine Bloom. Designed for use within Ableton Live, they offer a wealth of sampled Fairlight sounds: drums, keyboards, mallets, strings, guitars, basses, brass, winds, voices, effects and more are presented as ten different downloads. **sonicbloom.net**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STRANJAH
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Spitfire Audio

Hammers

$340

spitfireaudio.com

By Dave Gale

Strengths

+ Great sounding samples
+ Distinctive cinematic percussion palette
+ Looped content is highly effective
+ Warped section gives instant intensity
+ Real-time control elements are a boon

Limitations

- Limited selection of percussion sounds
- Loop section could be more substantial

In a new collaboration between Saw composer Charlie Clouser and Spitfire Audio, percussion has never sounded so brutal

Creating drums sounds for cinematic textures requires great drum samples, with a heavy layer of signal processing. This is where Spitfire's latest product could take the hard work out of your next project, with these unique hits and loops, driven to brutal distraction.

Charlie Clouser is a Hollywood composer best known for the angsty scores which accompany the not hugely heart-warming *Saw* film franchise. He actually started his musical life as a drummer, before becoming a keyboard player and programmer with the band Nine Inch Nails. Also known for their processed and dark musical tendencies, his simmering pot of stylistic percussion-based sound design led Spitfire Audio to his door, with a view to creating a software instrument to host percussion sounds for the current and next generation of cinematic composers.

Hammer flow

Hammers is based around Spitfire's very own virtual instrument, in a format that has become familiar. Instrument selection occurs at the very top, while realtime controls default to the upper part of the instrument GUI, where two familiar faders control expression and timbral/dynamic volume. The now-familiar Spitfire 'knob' can be assigned to one of five control elements, reverberation, low-pass filter, reverse, compression and normalize, although not all are available with all instruments. This section also allows the user to see what's occurring with articulation options, which may also be automated via key-press. There's also access to the numerous mic or bounced signals, for greater control of your initial sound construct. Spitfire has generated great overall stereo mixes, great for anyone running low on RAM; the more signal paths you load at once, the greater the RAM demands.

The Spitfire player often invites a conversation, in much the same way that T-bone-steak-flavored ice cream does. The player window is scalable, with an ability to hide or show controls. It's not generally a problem if working with a desktop screen, but it could easily eat up a laptop's screen space, annoyingly.

Initial instruments

While Hammers is exclusively percussion-based, there is great diversity in content color. The start point for the library is the initial recording of eight different drums or instrument types. These begin with bass drums and surdos, rising through the pitched ranks to toms, roto-toms, darbuka and frame drums. Snare drums and even a section titled Scrap Metal round it off.

The basic level of sample, of which there are 118,744 begins with individual hits and strikes. These include articulations of single hits, flams, ruffs and rolls. The default Ensemble patch is useful for loading a clean opening set, should you wish to trigger and program the sounds yourself. It actually makes an ideal place to begin building a groove, which is interestingly the way that Clouser likes to begin his own scoring work. By having access to so many instruments simultaneously, it bypasses one of the criticisms often leveled at Spitfire's door: the lack of multi-timbral operation within their own plugin. Granted, you won't be able to mix and match the articulations in this Ensemble setting, but you can at least get going, and with some clever programming, get around many issues.

> "Exudes cinematic color, providing the perfect backdrop for an energetic or pounding track"

For even greater control, loading individual drum instruments is an ideal route, at which stage you have access to solo hits, or hits with multiple players. Each drum gets different articulations; some use just sticks, while many use brushes, or indicate where the drum is being struck.

The darbuka is an excellent case in point. Originating from Egypt and Turkey, it provides a pleasant blend of high frequency slap from the skin/head, with plenty of hi-mid content from the body of the drum. The timbre shifts according to the strength of the strike, and location of hit, and with so many hits on offer, building a beautiful groove with a little help from your DAW feels very easy, intuitive and creative. There is a two-player option here too, beautifully imaged across the stereo image.

Clouser gets closer to the intricacies of drum recording then most. Sorry.

Get Real

Building a real-time drum groove can be a tricky business for anyone who doesn't have an understanding of how grooves can support thematic or harmonic material. It's one of the reasons that loop libraries can be so appealing and useful, not to mention time saving.

Building up a complete groove using Hammers is a very desirable and enjoyable musical pursuit, but it can take time to get something coherent and believable, if building your own rhythmic passages. Hammers arguably takes the hard work out of the process, thanks to the looped content, which can be nicely mixed and matched, to suit your purposes. If you're trying to create a sense of build and progression using Hammers, the real-time control elements are invaluable. The ability to close-off the filter, limiting Hammers to its low end content, creates another quick and effective route to builds. Use alongside a third-party filter, which offers control of resonance, and it's possible to create even more tension.

This can also be an excellent way of masking any inferiority which you may or may not have in your own groove programming. Smoke and mirrors are always helpful bedfellows in production where the Holy Grail is a believable human facsimile.

Warped!

Accompanying the individual hits is a compendium of live loops, all played and constructed by Clouser, with two further session players. The loops are inspiring, inviting the user to explore the real-time controls for driving energy, or upping the feel with effects.

For more acoustic work, these loops are a real boon, and can easily be treated within the DAW, in order to create darker colors, much like the sounds that Charlie Clouser is renowned for. However, if you lack confidence in this pursuit, Hammers comes quickly to your aid, with a large collection of Warped loops, which take on that Clouser identity. These exude cinematic color, providing the perfect backdrop for an energetic or pounding track.

Hammers is a great collection of drum and percussion sounds, offering single articulations and some excellent looped content. It is limited in overall percussive color, but offers a unique and distinguished palette which is more left-field.

What you do end up with here is an excellent package of great sounds, but we feel that you'll extend the life cycle of Hammers by placing it alongside other percussion or drum packages. ■

THE ALTERNATIVES

SONIVOX
Big Bang Cinematic
$199
Brimming with huge wallops, perfect for your cinematic tracks

OT
Tom Holkenborg's Percussion
$345
Great sounding drums includes plenty of big drums, including marching band-style ensembles


FOCAL
AKAI
FOCAL

Focal

Alpha 65 Evo

$814 per pair
focal.com

By Jon Musgrave

Strengths

+ Open detailed sound with broad sweet spot and cohesive low end
+ New Slatefiber woofer
+ Contoured tweeter waveguide
+ Upgraded cabinet styling
+ Flexible onboard EQ

Limitations

- Wide footprint

If you're looking to move up from your budget monitors this pair could be just the thing you need

Focal's more affordable Alpha monitor range has had an upgrade and been rebadged as Alpha Evo. Currently available with either a 5" or 6.5" woofer, it's the larger Alpha 65 Evo I have on the test bench.

The first thing to say about these monitors is they look fantastic. The styling has been overhauled, and you now have smoother corners and a single front-facing slotted bass port. The tweeter waveguide now offers a more graded profile for the 1"-inverted aluminum dome tweeter. Other improvements include a new woofer which uses Focal's new 'Slatefiber' cone material, Class D rather than Class AB onboard amplification, and a choice of three inputs – RCA, XLR and TRS.

The cabinet styling does give the Alpha 65 Evo quite a wide footprint, however at 7.6kg they're not as heavy as they look. This makes them ideal for wall or ceiling mounting and on the back you'll find suitable mounting points. Audio hookup is simple enough and you can connect both the balanced and unbalanced inputs simultaneously, which is handy, although note that the TRS jack does override the XLR. There's no level control, which is a shame, however round the back you'll find a switchable sensitivity (0 or +6dB), providing some flexibility. On the back panel you'll also find a switch for the auto standby mode (on/off).

Much like the original Alpha 65, the Evo has good onboard EQ. The two shelving filters offer +/-3dB at 4.5kHz and an impressive +/-6dB at 300Hz, and with smooth rather than notched gain you can get the precise settings you want. They also sound very gentle even at extreme settings, so you can't go too far wrong, though more surgical EQ is not an option.

Sonically the Alpha 65 Evo sounds bright and open with excellent space and separation, although I would say it's not particularly forward in the mid range. And the sweet spot is pretty broad both vertically and horizontally, which I like. On initial listening I did find them quite lively in the higher frequencies. Focal advises modifying the HF EQ to match the room acoustics and although my room is not particularly reverberant I found -1.5dB resolved this issue pretty well. That said, I would conclude that the monitors are most definitely on the brighter side.

In the low end I had no issues at all and the front facing ports gel really well with the LF drivers to deliver solid, cohesive lower frequencies. The low-end extension is also pretty good with frequencies audible down to 40Hz. In fact, I think the low end is my favorite aspect of their tonality, and EQing kicks and bass sounds I found suitably revealing.

Overall these are classy monitors, though at over $800 a pair, on average, they're likely to be pricier than their predecessors, and no small change. That said, they are very revealing in use, and I think you're getting a monitor that is pretty close to considerably more expensive professional designs. ■

M-AUDIO
OXYGEN PRO MINI
PAD BANK
NOTE REPEAT
LATCH
TEMPO
BACK
TYPE
PAN
OCTAVE
DEVICE
GATE
SENDS
SWING
ARP
DAW
1/4
1/8
1/16
1/32
1,3,5
1,3,7
1,3,5,7
1,5,9
1,5,12
3,5,1
5,1,3
RAND
MAJOR
MINOR
CUSTOM
GLOBAL
MODE
SCALE
CHORD
LATCH
BANK <
BANK >
SHIFT

M-Audio

Oxygen Pro Mini

$136

m-audio.com

By Jon Musgrave

Strengths

+ Quality RGB backlit performance pads
+ Good range of onboard controls for DAW control and editing
+ Onboard or remote editing of controller assignments

Limitations

- No aftertouch
- Buttons are a bit noisy

A quality mini-key keyboard coupled with plenty of performance features should be a winning combo

The Oxygen Pro Mini is the smallest of the four Oxygen Pro keyboards and the only one that features mini keys. At 15.5ins by 7.5ins and just over 2lb, Oxygen Pro Mini is also pretty portable. Much like their Keystation Mini you've got 32 keys, which is a decent number for a compact design and handy for composition. And I have to say, although I'm not a massive fan of mini keys in general, the PrecisionTouch semi-weighted action feels great. The only slight disappointment: no aftertouch.

In addition to USB and sustain pedal connections, there's a MIDI output. This is on a mini-jack and to use this you'll need a converter cable which, alas, is not included. However, once hooked up, you can set the MIDI out to be sourced from either the keyboard, DAW or both. So the Pro Mini also doubles as a MIDI output interface for your external MIDI gear.

One of the great things about the Oxygen Pro range is the extensive performance controls and Pro Mini doesn't disappoint. The 8 RGB backlit velocity sensitive pads not only feel great but also feature handy options like momentary and latching Note Repeat. You've also got two pad banks, doubling the number of available pad locations. The assignable faders, knobs and buttons (four of each) are mapped according to the onboard DAW presets, making setting up a breeze. You've also got six dedicated transport controls. Many buttons are backlit, which is handy, and feel precise, albeit a little too noisy when pressed.

Pro Mini operates in one of two distinct modes – DAW and Preset – which control DAW and softsynths/plugins respectively and you switch between them using the dedicated button. Presets can be edited extensively onboard using the push button rotary encoder and OLED screen. However, there's also a downloadable software editor, and though this uses a more long-winded send/receive process, I found that for more extensive changes this was quicker and more intuitive. Oxygen Pro Mini comes with a decent software bundle including Ableton Live Lite, Pro Tools First M-Audio Edition, MPC Beats and a number of AIR soft synths, so it's no surprise that there are a number of ready-made mapping presets for AIR synths (Hybrid, Velvet, Xpand! 2 and so on).

Oxygen Pro Mini has some further handy performance options including an onboard arpeggiator with range, gate and swing settings and two Smart options – Chord and Scale. The latter is particularly cool if you're not a great keyboardist as you can set your key and scale type, then if you hit any notes outside the key, they will play to the nearest scale note.

Overall Oxygen Pro Mini has an impressive feature set, is easy to use and feels solid and professional, and despite some minor gripes, at $136 this is an absolute bargain. ■

REVIEW

Softube

Model 84

$183
softube.com

By Andy Jones

Strengths

+ Cool-sounding facsimile
+ The GUI is really inviting
+ Great emphasis placed on accuracy
+ The Chorus is nice and clean
+ EQ Bass boost does the job brilliantly!

Limitations

- Lacking embellishments
- No standalone version

As another software rebirth of a vintage classic from 1984 arrives, Softube delves into Greek mythology (via Japan) for inspiration

Back in the 1980s, Roland had the good sense to do a really smart thing. The late '70s poly-synth behemoths, which in many cases required hair of a similar magnitude, were big, bloated and expensive. This provoked a reaction from Roland to produce a more affordable polysynth, in part thanks to the shrinking of electronic components, that made a sensible price point more obtainable. And so, the path was laid for the future classics which came from the Juno Series.

With reference to Greek mythology, Juno was the wife of Jupiter, better known among mere synth-mortals as the flagship synthesizers in the Roland range. Anyway, Roland created the Juno 6; a six-note polyphonic synth, now recognized as a vintage classic. Within a matter of months, the Juno 60 appeared. In essence, this was the same synth but with added patch memories and a connectivity format called DCB (Digital Communication Buss) which allowed easy connection to Roland sequencers et al. Analog issues aside, those earlier Junos sounded rich and deep, in part thanks to the rather noisy chorus circuit, placed on the backend of the signal. The premise being that with only one oscillator per voice on board, the chorus would help thicken the sound, and so it did!

Fast-forward, then, to 1984, and the third coming of the Juno arrived in the shape of the 106. Gone were the wooden end-cheeks, arpeggiator and DCB. In came portamento (glide) and a new protocol called MIDI, in one of its first fully-fitted deployments.

At the time, Roland was going to quite considerable efforts to try and prove that the analog stylings of the 106 could compete with the new great pretender from Yamaha, called the DX7. Hence the sound of the 106 was tighter, sharper and crisper. The argument about which Juno sounds better is one that we'll save for another day, but it is nevertheless significant that the 106 has its notable fans, for punchy synth lines and weighty basses, within the commercial and production psyche today. This might explain why Softube has zoned its 24k magic on the 106, in much the same way that the recently applauded Model 72 provides a MiniMoog re-born.

Boogie with a suitcase!

The fashionable architecture of the Juno is carried forward to the Model 84, through a familiar fascia. With reverence to most second hand originals, the paint work appears chipped and scratched, with a couple of lumps taken out of the rear edge, presumably where it's been slammed up against its 'virtual' X-frame keyboard stand. It looks really cool and engaging, but more importantly it sounds über cool, with the DCOs presenting an almost perfect match to the original harmonic makeup. The early Juno synth range used a form of oscillator which was certainly overtly analog, but placed under digital control. Hence they were known as DCOs (rather than VCOs), producing more reliable tuning stability, unlike their synthesizer cousins from some years earlier.

As this is a virtual facsimile, Softube has sought to follow through the 106 thread, with buttons allowing the activation of the fundamental saw and square waves. There is no volume control for either, but there is a fader to control the sub-oscillator volume. This is pretty essential, as the sub can quickly obscure the fundamental, while associated with more polyphonic patching.

Pulse Width Modulation (PWM) is applied to the square/pulse wave, either through manual control to LFO modulation. The earlier Junos also allowed for envelope control of PWM, and while it's easy to get dragged along with the attention to 106 detail, that's one addition that would've been a welcome reintroduction for the Model 84, although Softube has implemented an additional control panel, which springs open from the right of the window, allowing PWM control via velocity and aftertouch.

Filters and envelopes

The filter, being a 4-pole design, sounds incredibly exacting. Softube has even recreated the obvious stepping, exaggerated in self oscillation, which was a recognizable trait for the 106. The resultant sine sounds incredibly accurate, compared to the original. Softube have slightly repurposed the High-pass filter, changing it by way of description to EQ. It offers two levels of high pass filtering along with a bass boost, which is handy when using the Unison mode, which stacks all six voices for a great bass texture. It's also possible to detune these, using a fader from the 'spring-out' control panel.

> "Sounds über cool, with the DCOs presenting an almost perfect match to the original harmonic makeup"

Gaining control

One oddity with Softube softsynths is the lack of standalone operation. It's sometimes quite nice to load up a synth on-screen and get lost in playing, rather than getting bogged down with the DAW. However, Softube have made the process of assigning a MIDI CC to a plugin fader very easy, so much so, it invites the user to get extremely hands on with a suitable hardware controller. You could also employ one of the many assignable MIDI controllers for iPads/tablets, building a template with the relevant MIDI CCs, for a more Juno-istic experience. Of course, this moves immediately into focus when hooked in with your DAW, as is Softube's intention, where the ability to control all elements in realtime bring the Model 84 to life. It's so easy to introduce a filter swell or increase amounts of PWM, as your track builds, that it could really set apart your sound. Let's face it, this is a Juno 106 in plugin form, and the only person who'll know that it's not a real Juno is you!

Juno envelopes were never the strongest designs on the originals. They were useable enough, but never the snappiest, and this transcends to the Model 84. If we're honest, the shape and behavior feels less convincing here, with extended phases too. This isn't going to impact hugely on usability, and wouldn't be noticeable unless placed in direct line against an original, but it's worth noting for purity.

The final output

One crucial element of the Juno's architecture is the chorus, which unlike the original is clean, effective and displays enormous reverence to the original. For just about all purposes, the Model 84 sounds just like a 106, with all of the associated conveniences of a plugin, such as the ability to easily apply a hardware MIDI CC controller to a pot or fader, allowing simple editing within the DAW, either on-the-fly, or under DAW control.

Sonically it delivers well, but it does feel like a slightly missed opportunity which invites the return of certain much-missed elements or enhancements; envelope control of PWM, an arpeggiator, and while we're on this thread, access to a second envelope would have been nice too, along with the ability to extend the voice count beyond six.

Don't get us wrong, it's a great sounding synthesizer and a huge bargain against a hardware original, but we're nearly 40 years on, and extra bells and whistles would go a long way to enhancing the DAW-based Juno experience. ■

THE ALTERNATIVES

ROLAND
Juno 106
$149 (or as part of Roland Cloud)
Truly authentic emulation, with some tasty enhancements

ARTURIA
Juno 6v
$230
Close to the original Juno 6, but with added patch capability and other modernizing enhancements


AKAI
PROFESSIONAL
MPC ONE
Plugin 001
VEL > AMP
0 %
MW VIB
100 %
AT VIB
0 %
VIB SPEED
0.59 Hz
KEY ON
KEY OFF
Mellotron
FORMANT
SMPL START
0 %
CUTOFF
100%
ATTACK
RELEASE
MODEL
FLAVOR/COMP/EQ
DELAY/SPRING





Akai

MPC 2.10 update

Free
akaipro.com

By Simon Arblaster

Strengths

+ Great new effects and plugin instruments
+ Audio interface support
+ It's free!

Limitations

- Small screen real estate means much menu scrolling

Does this free major update to the platform improve the MPC hardware? Let's get stuck in to find out

Akai's 2.10 firmware update for its MPC range is one of the most significant updates to the platform yet, with a slew of new features that promise to add even more value to the MPC hardware range. Four new synth plugins have arrived: three are recreations of the classic Solina string synth, Mellotron and indomitable Odyssey, and an entirely new instrument called Hype rounds off the list. A new Vocal Suite includes three insert effects bringing you automatic tuning with the Vocal Tune, four-part harmonization via Vocal Harmonizer and a doubling effect.

Alongside the Vocal Suite are four new and one updated insert effect, from Air Technology and two new insert effects from Akai. The new offerings from Air are Half Speed, Stutter, Diode Splitter and Limiter, while Diffuser Delay gets updated to include Low Cut, Width, Sync and Pan parameters to an already useful effect. The new Akai effects are Granulator and Sample Delay.

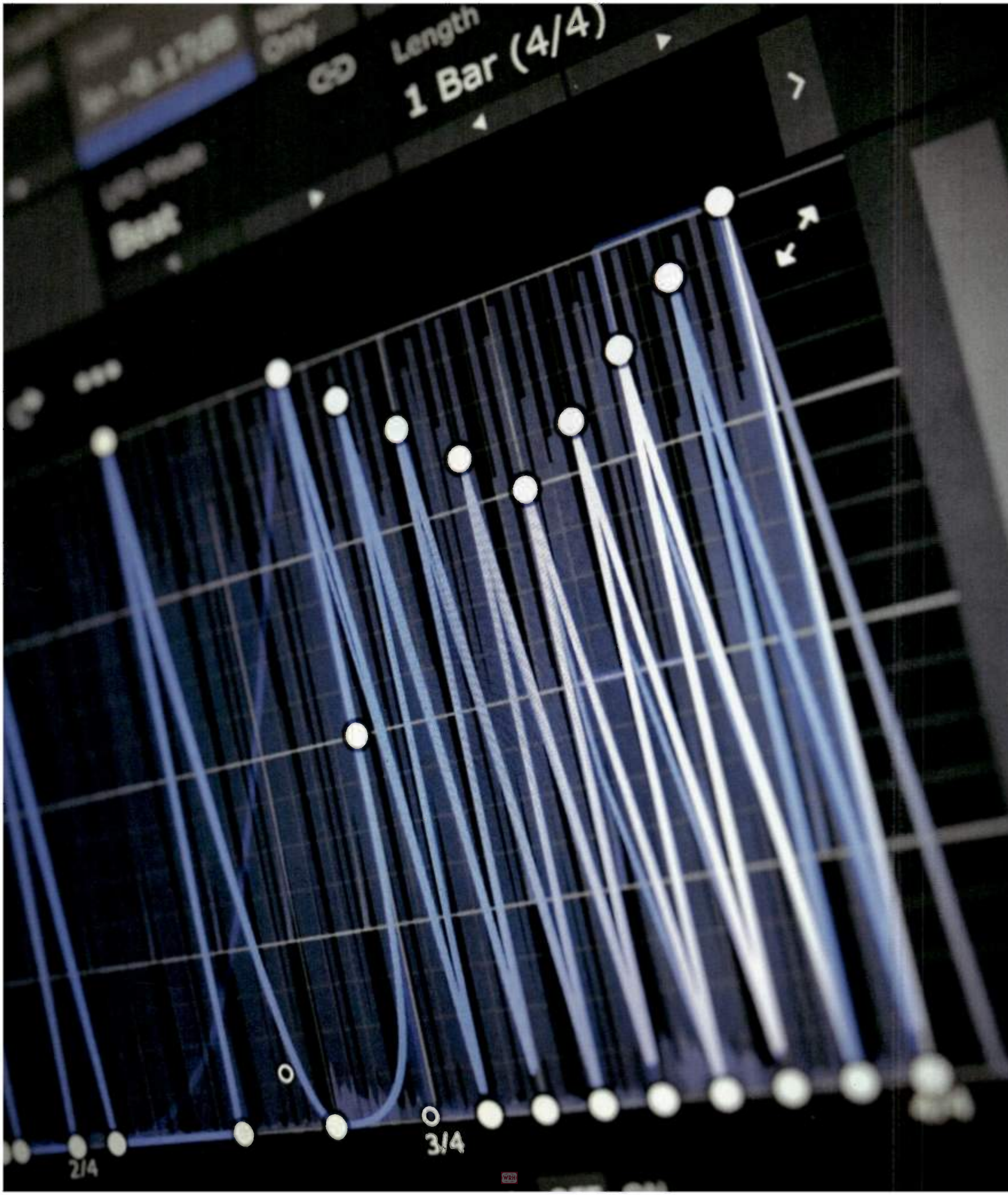
Perhaps the biggest boon is that 2.10 now offers support to host any class-compliant USB audio interface, for a max 32 inputs and outputs.

As updates go, we'd have been content with just one plugin instrument, but having four is spoiling us. We have three classic synths emulated that offer practically everything you need. The Odyssey, being a particular favorite, is an absolute must for any rasping bass layers and this version improves on the old recipe by adding polyphony. The Solina and Mellotron both deliver strings and solo instruments which blend so well when dropping lo-fi and retro beats. These aren't just faithful recreations, but each also adds in its own effects, vastly improving on the original. You can add extra grit to the Mellotron with tape-based effects, while the Solina comes equipped with delay, reverb and chorus.

It's not all about the retro vibes as the biggest synth available in the update comes in the form of Hype, which features multiple synth engines: wavetable, FM, virtual analog, and sampled. It has a wide array of effects and presets to get you started before you even need to start digging below the surface.

Hype is very much designed with the MPC macro controls in mind, so you'll find tweaking much more hands-on, especially compared to the three retro-synths, which have been designed, admirably, with the small screen real estate in mind. Adding four plugin instruments, the new Vocal Suite, plus other insert effects would be enough for some folks to stay in the MPC box without the need to reach for the nearest DAW. Throw in the USB audio interface support and it makes the latest iteration of MPC a must-have.

So does this free update to the platform improve the MPC hardware? In a nutshell, yes it does. As free updates go, MPC 2.10 is jam-packed with even more tools to keep you in MPC ecosystem. ■

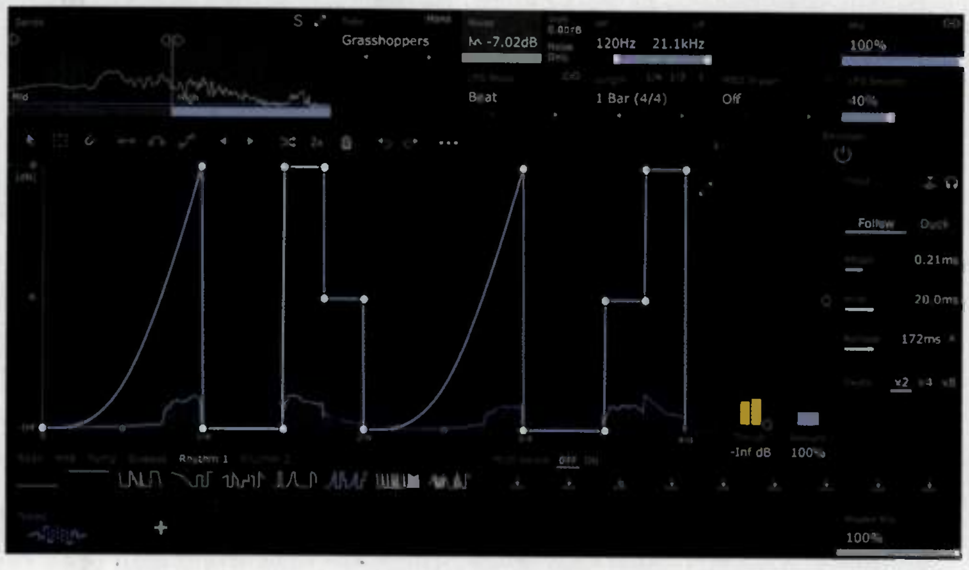

Length
1 Bar (4/4)
2/4
3/4

Cableguys

NoiseShaper

$56

cableguys.com



By Si Truss

Strengths

+ Fantastic range of source noise samples
+ Lots of rhythmic modulation tools
+ Complements other Shaper effects well

Limitations

- Multiband setup can be confusing

ShaperBox branches out from effects processing with a sample-powered noise generator.

Regular readers will be aware of our fondness for Cableguys' ShaperBox 2 – the plugin which acts as a host for a library of Shaper effects, which now include the latest addition to the package: NoiseShaper.

NoiseShaper differs slightly from the other effects. Instead of applying an effect to the incoming audio, NoiseShaper creates sound itself. As you might have guessed, the primary focus here is noise. It comes with a library of noise samples, which can be blended with the incoming audio and modulated in a variety of ways to create dynamic, rhythmic effects.

Much of the familiar ShaperBox workflow remains unchanged. The primary element of the UI remains the titular custom modulation shaper, which allows users to craft a wide variety of loopable envelope shapes or pick premade ones from an array of templates. As with other Shapers, the incoming signal can be split into three independent frequency bands, each with its own parameter and modulation settings. The plugin also offers an assortment of MIDI triggering and switching tools – great for live jamming and performance.

For each band, NoiseShaper offers a dropdown to pick a source sample from the library. These range from simple white or pink noise loops to more interesting textures, such as broken equipment, field recordings or vinyl crackle samples. The range and quality is great. Users can also use their own loops by dropping them into NoiseShaper's sample folder, although the range on offer is good enough as is.

The envelope follower – a familiar feature – takes a front seat here. NoiseShaper works best when the noise samples pump in time with, or around, the incoming audio signal, so it makes sense that the follower is always on by default.

One odd element of the design is how it interacts with the multiband splitter. As with other Shapers, the incoming signal can be split into three bands, but here the noise samples are always applied with their full frequency range. I understand the reasoning here, as it allows users to, say, have high-frequency tape noise respond to just the low-frequency pulse from a kick, but the design also feels at odds with the multiband nature of ShaperBox. Admittedly, there's an additional high/low filter that can be applied to the noise samples anyway, but it would be nice to have an option to engage/disengage whether each band's noise is affected by the multiband filter.

That aside, NoiseShaper is fun and has the potential to be the most creatively powerful tool in ShaperBox's arsenal. It's great for creating builds or adding drama. Some of the most fun applications come from layering multiple lines of rhythmically pulsing noise to create complex grooves. A must-try for existing ShaperBox users, and yet another argument why the uninitiated should take it for a spin. ■

KV331

SynthMaster 2 iOS

$13.50
synthmaster.com

By Andy Price

Strengths

+ Awesome modulation effects
+ Flexible UI options
+ Perfect for sound design on the move
+ Great value for the price

Limitations

- Steep learning curve
- At times hard to control multiple rotaries

Bringing the fully-featured flagship synth to iPad/Phone users, is this breakout version of KV331's creative powerhouse the master of all?

We've long been tinkerers with SynthMaster One, KV331's 16-voice semi-modular beauty that first appeared in our App Store in 2018. While we were dazzled at the high-quality wavetables and flexible sound design potential that this ridiculously underpriced monster provided, a truly mobile version of the more versatile desktop version of SynthMaster would be a much more appealing prospect. Now, with the desktop SynthMaster maintaining its status as a renowned player in the softsynth hierarchy, KV331 Audio has released the iOS successor. This time, this isn't a 'lite' version. SynthMaster 2 sports all the bells and whistles of its desktop parent.

The main headline here is that KV331 has essentially ported SynthMaster 2.9 – in its feature-packed entirety – to the iPad. First off there's the robust sound engine, which houses the virtual oscillators, modulators and filters. The app's synth stages are programmable via a slickly re-tooled, cross-platform UI which also supports the native resolutions of numerous iPad types. We're also given 1,000 top tier presets as standard, designed by such luminaries as Ubukata, NatLife and Yuli Yolo – a notable increase on SynthMaster One's 500.

Navigation through the browser is fluid, and before too long, the sonic magnitude of SynthMaster 2 is revealed. Across diverse presets such as the retro-vibed cool of FM Strings MS, the percussive impacts of BAS Aluminum or the dirtied up synth-basses, it's clear this app can contend with the likes of Arturia's Pigments and the other big boys of synth sound design.

The primarily rotary controls are simple and you'll be on familiar ground if you've used any of the previous iterations. The softsynth's engine is built around its dual-layer oscillators, which can be set to different wave-generating approaches. Basic provides single-cycle waveforms or samples, while Additive allows eight oscillators to be stacked up, resulting in beefy tones, Wavetable mode allows for more gradual and nuanced waveshape blending, and Vector mode mixes four oscillators in a 3D, spatial audio-leaning way.

Next up is a supremely detailed raft of modulators and filters which can muddy, detune or hammer your sounds into shape. You can also view your sound-chain with real-time visual feedback, giving the user a better grasp of how sound is being effected – editing in this section is all the more simple, too. The depth here is staggering, inviting you to experiment with each encoder to see how far you can go.

Our only issue was making those quick, subtle rotary tweaks as we performed – often they wouldn't detect that they had been touched, though perhaps this is more a flaw with the iPad.

It's clear from our early experiments that SynthMaster 2 will be our new go-to for quick synth-idea generation. Every aspect of the already top-notch experience has been refined to complete the story as it always was meant to be. ■

REVIEW

UVI

Super-7

$91

uvi.net

By Dave Gale

Strengths

+ Charming representation of an unloved classic
+ Superb, production-ready sounds
+ Laden with Juno-106 content
+ The drum section is very comprehensive
+ A bargain, given the amount of content

Limitations

- No onboard drum sequencing included
- Sample based, not modeled

What happens if you weave a karaoke classic into a programmable sample-based instrument? The results from UVI will be Super!

The MKS-7, also branded the Super Quartet, was a desktop and rack-mounted band-in-a-box, split into four sections providing bass, chords (synth), melody (lead) and drums. As it was preset, editing was severely limited. This was a colossal shame as the synth engine was borrowed from the Juno 106, while the drum sounds were similarly repurposed from the TR-707. Shortly after release, the MKS-7 completely bombed in price, leaving it to the tech-savvy to infiltrate the MKS-7 with SysEx editors, allowing control of the onboard sound sources.

Thankfully, UVI has an extensive and highly enviable track record in sampling modules of this kind, which it has undertaken with the usual panache and detail. Running within the developer's reliable, freely available Workstation player (or Falcon 2 synth), the Super-7 is split into four segments, like the original.

Beginning with the drum section, UVI has included the 707 complement, but also added CR-78, 606, 626, 808 and 909 samples. Also included are elements from their excellent Drum Designer suite. It's worth celebrating what we have here; the 707 has become a cult classic, with a sampled sound more akin to a Linn Drum, with additional representations from the ever-analog elements of the 808/909. Moreover, these UVI sounds are production ready, with solid weight and spirit. Simply fantastic. Somewhat regrettably, there is no drum pattern editor, but you can trigger preset patterns, while also triggering individual sounds from within your DAW. It's also a simple procedure to export the patterns as MIDI files, for dropping into the DAW and programming potential.

Over in 106-land, preset sounds are split between the Bass, Melody and Synth sections. While there's similarity between these areas, the distinction is apparent through the preset content and its access to polyphonic operation.

The bass sounds benefit from that beautifully familiar 106 depth, and while there are a smattering of presets, UVI has sampled a huge number of 106 waveforms, combining Saw, Square, a sum of both and the sub. These are great startpoints for user patches, alongside the section's arpeggiator. Meanwhile, the fantastic overdrive and equalization sections shore up those UVI production-ready principals.

Let's applaud UVI; the Super Quartet was a mostly-failed box with great charm. UVI has captured this spirit, providing multi-layered-presets for an instant '80s karaoke soundtrack, with a single note trigger. The Super-7 gives all the 106-style samples from the ground up, with an additional multimode filter, which provides a versatile color as an addition to the sampled content. Add extensive effects, two envelopes and tons of programmable options and this is a nifty way to get a 106 and 707. ■


KRK

KRK

S10.4 subwoofer

$477
krkmusic.com

By Jon Musgrave

Strengths

+ Balanced and unbalanced inputs
+ Good bass extension
+ Foot switch bypass option
+ Flexible crossover settings

Limitations

- Reasonably large enclosure
- No dedicated LFE input

Looking for some extra bass in your monitoring system? Fire up this latest yellow-coned beast

Monitoring the lowest frequencies in your productions can be tricky and is influenced by factors including your workspace and your monitors. Most of us don't have the space to accommodate large monitors with an extended frequency range and often use workarounds such as frequency analyzers or headphones to get a better feel for the low end. However, one option is to add a sub bass unit to your monitors.

Choosing a sub that fits the bill often boils down to the features on offer, size and of course price. As you may have deduced, the S10.4 is built around a 10" driver, and this makes it their mid-size design, between the S8.4 (8") and S12.4 (12"). It's worth saying that KRK also produce another more high-end and considerably more expensive 12" sub, the 12sHO.

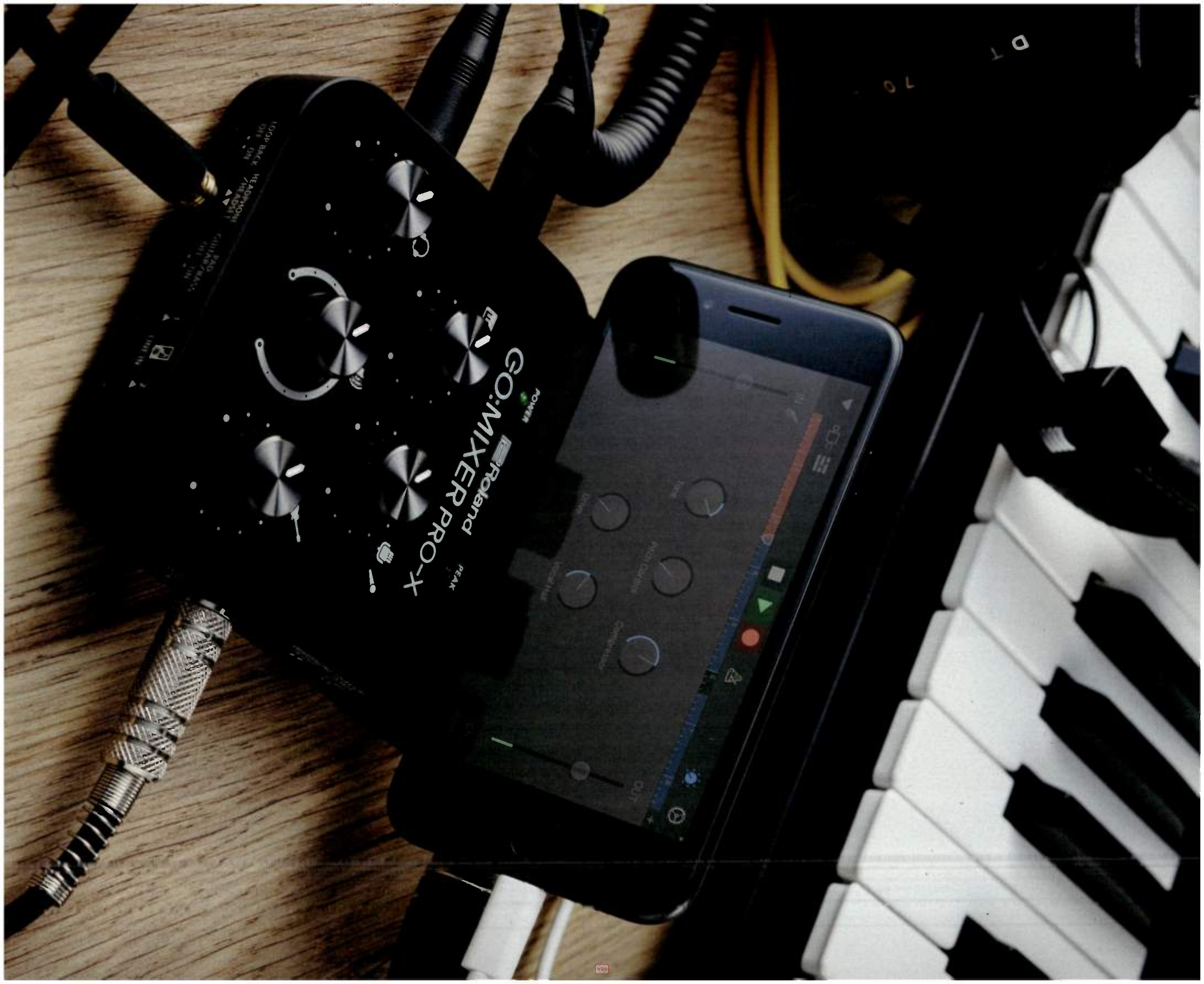
The S10.4 uses a front fitted 10" glass aramid composite driver with KRK's distinctive yellow color scheme. Low frequency delivery is further enhanced by a front-facing slotted port and if you position the sub on its longer side, as intended, the port is vertical and up the right-hand side. Either way, at almost 19ins wide the S10.4 is still a reasonable size to accommodate.

The S10.4 is designed to be used in line with monitors and, to that end, on the back you'll find stereo inputs and outputs on both unbalanced (RCA) and balanced (XLR and TRS) connectors. There's no separate LFE input, which you would use with an external monitor controller system.

Rear mounted controls are Volume, Input Sensitivity (Normal/High), Polarity (0 or 180 degrees), Ground lift (on/off), auto Standby (on/off) and the all important Crossover. This has four notched settings (60, 70, 80 and 90Hz) allowing you to tailor the sub to your monitor frequency response. There's also a footswitch input and when engaged this bypasses the sub and the filter so your monitors receive the full frequency signal. You can use any generic latching footswitch, and though this is an additional expense, it's a vital addition so that you can easily switch between monitoring with or without the sub engaged.

KRK's manual includes helpful setup and placement info and the advice is to keep the sub close to your monitors and away from room corners. Further, more detailed, advice includes using pink noise and a decibel meter to set levels and polarity. Nevertheless, positioning a sub does often require trial and error and when I positioned the S10.4 where my existing sub is, I found the polarity-inverted setting delivered the best results. The S10.4's Class D amplifier delivers plenty of punch. With 160 Watts to play with and a maximum peak SPL of around 117dB you can generate shuddering frequencies, and in testing it handled 30Hz very well.

A well engineered sub, with a quality finish and setup. Follow KRK's advice and you'll achieve a reliable low frequency extension of your monitoring system. ■


Roland
GO:MIXER PRO-X
POWER
PEAK

Roland

GO:MIXER Pro X

$150

roland.com

By Simon Arblaster

Strengths

+ Better smartphone compatibility
+ New guitar/bass pad switch

Limitations

- Could do with a sturdier chassis

Built for video creators and portable recording, does the latest GO:Mixer Pro have the X factor?

The premise behind Roland's GO:MIXER series is simple: to give your smartphone as much audio I/O for every musical eventuality with video creation very much at its heart. The latest version, the Pro X, retains the same form factor as the two previous generations – clearly, Roland believes this is a winner – but adds some much-needed tweaks to compatibility.

For the uninitiated, the Roland GO:MIXER Pro X features 11 audio input channels (two more than the Pro) and three output channels (one more than the Pro). This is configured through two (L/mono, R) 1/4 inch jack inputs, two 3.5mm stereo line inputs, 1/4 inch Guitar/Bass input, a TRRS smartphone In/Out (stereo, CTIA) and a combo XLR/1/4 jack input. There's also a headphone or headset jack (stereo, CTIA) and the obligatory micro USB socket to round off the I/O.

The unit can be powered by three AAA batteries, or straight from a host device, such as your smartphone or tablet and Roland gains extra points for supplying the appropriate cables, however, batteries are not included.

Five knobs adorn the top of the unit for control over the guitar/bass input, line inputs, mic input, headset input and an overall output level.

As we've already mentioned, the three extra ins and outs are an improvement on the Pro, with the inclusion of the TRRS port, meaning that you can use a headphone mic or headset and the Smartphone I/O opens up use on any device with a TRRS connection. There's also the much needed guitar/bass pad to help tame any high levels from active pickups etc. Further switching on the unit includes phantom power, on/off and the Loop Back function.

Switching off the Loop Back function is ideal when you just want to monitor a backing track whilst recording, so if you're tracking vocals you will just capture the vocals. The alternative is to utilize the Loop Back function to record in a live scenario, including any arrangements or backing tracks from your smartphone. With the announcement of the Pro X we had hoped that multi-channel audio support might have been included, but sadly not, so all the channels are mixed into stereo. It's by no means a real issue as the Pro X still performs admirably at the job of mixing multiple sources in multiple scenarios. Maybe it should've been referred to as the mk2?

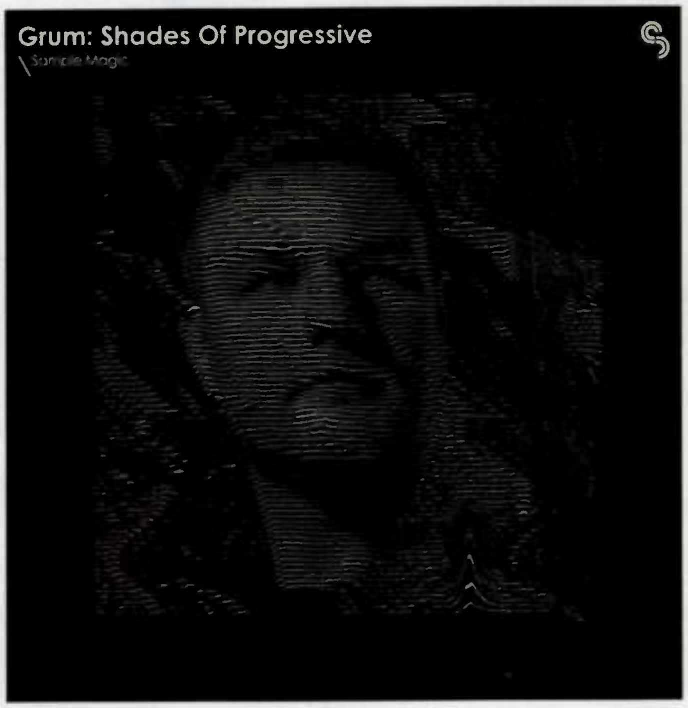
The form factor hasn't changed since the original, which is no bad thing really. But, if you're using the smartphone/tablet cradle for shooting video and have everything plugged into the unit, it's worth noting that it's easy to end up dragging it around by heavy cabling. A sturdier chassis here would be more advantageous. You can forget shooting in portrait for TikTok, too, unless you have your own accessories to do so. You can use the Pro X with any camera or audio app, but Roland obviously pushes its own.

The GO:MIXER Pro X is one of the most feature-rich smartphone mixer/audio interfaces out there for video creators. The new X version vastly improves on its predecessor despite what seem like only minor tweaks. ■

Sample Magic

Grum: Shades of Progressive

From $7.99/month



Forward-thinking loops, one-shots, and presets from the Scottish house and trance producer. And the 300+ files on offer cover all the bass and bases you'll need to craft a beat that bangs like this fella right here.

As highlighted on his last three long players, Grum likes to dabble in cinematic soundscapes, deep and dark grooves, and ambient rises and falls. And he uses this opportunity to lay bare the elements that help define his moody and uplifting take on dance music.

Punchy drum hits, clubby lead lines, and stone cold low-end business, fills out the collection. All recorded through a high-end reel-to-reel tape machine for an extra layer of warmth and sheen, often lacking in other packs in this field.

Roy Spencer

splice.com

Audiotent

Operators Arturia DX7 Presets

$50

Preset pack for Arturia's sterling re-imagining of Yamaha's mid-'80s powerhouse DX7 synth. More booster rockets on the already iconic sounds, here. With extra attention paid to the expressiveness of the tonality, while increasing the already stunning breadth of timbres on this iconic keyboard.

Super deep sonics are at your fingertips, via the four macros and mod-wheel assigned presets. Every bleep, bass, modulation, and pulsing note sequence should inspire creativity, while lending high-end body, and richness to your productions. To use the presets you need Arturia's DX7 VST. But, that's just more money well spent.

audiotent.com

Loopmasters

Urban Agency Drum & Bass Vol 1

$42

Micky Finn's Urban Agency is brimming with talented DJs and producers. So what better way to showcase that, than with a new series of sample packs?

The inaugural offering sees four heavyweight drum & bassers set the stall: Voltage, Heist, DJ Limited, and Original Sin. It's a killer cavalcade of sounds from each player. Heist's drums are as good as any he put on Full Cycle or Metalheadz. Voltage's bassline crates once again show why he's the king of the rollers.

Everyone else steps up, covering a wide selection of piano loops, hats and percussion grooves, FX, sampler patches, and much more.

loopmasters.com

Samplestar

Slow 80s RnB Vol. 2

From $7.99/month

Syrupy and screwed samples. Lit with a woozy neon glow, and damp tape-drenched layers. Heavy on the melodics, and skittering hats. Everything beefed with that golden era's analog production presence.

Grab a polysynth from the cooler, and wash it down with a monstrously processed Thriller snare line, and you've got something worth the flashbacks.

Elsewhere, bubbly Zapp basslines strut, and Princely purple claps and drum hits rain down. It might say '80s on the packet, but these loops are ripe for any neo-soul, glitchy funk, and trappy LA-centric beat. Or Weeknd warriors still chasing that After Hours retro synth high.

splice.com

Blast & Loopmasters

Ska Horns

$34

Maximum honkage in this monster pack of brasstastic loops and riffs.

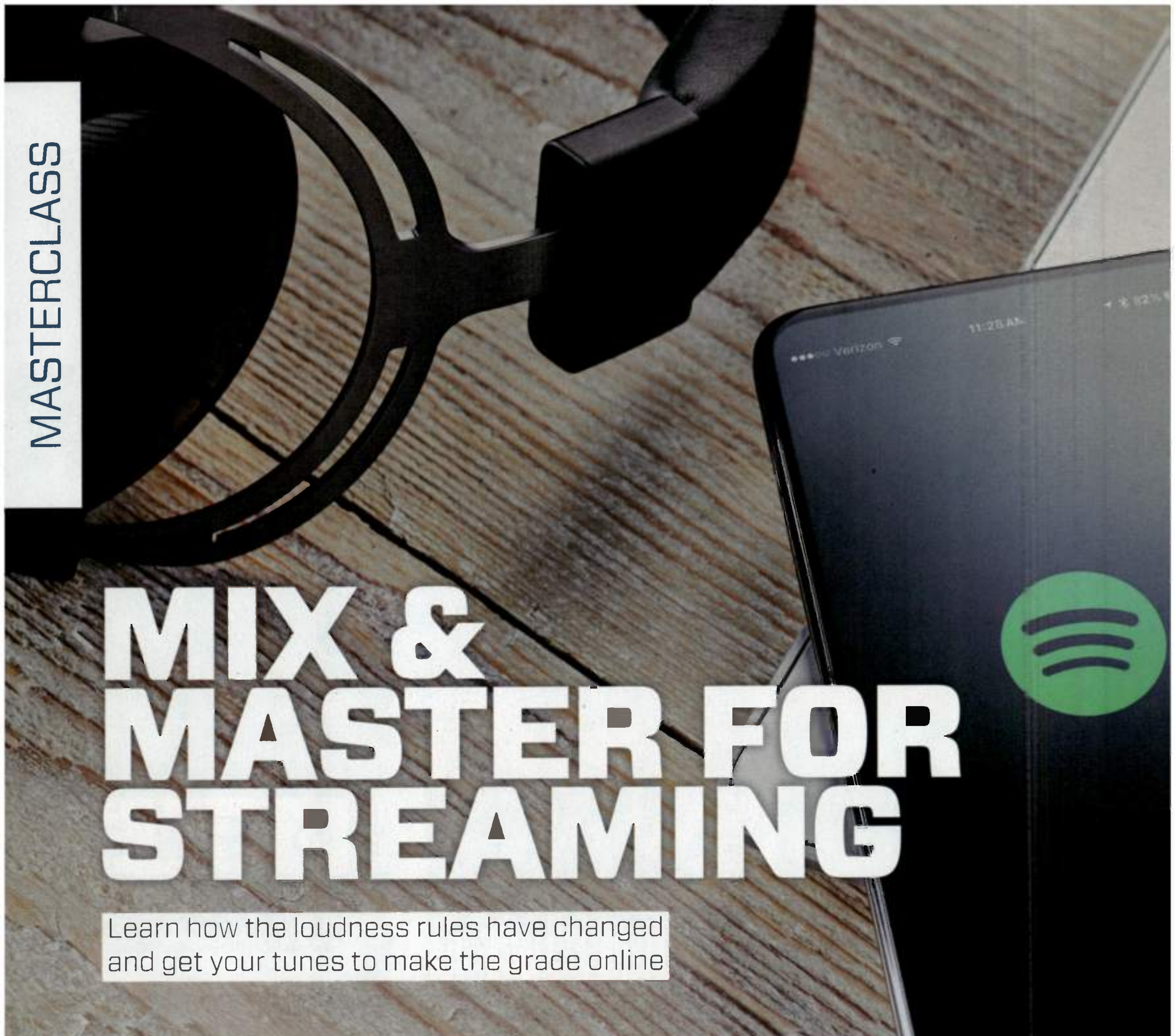
The guys locked in the studio for this one can really blow with the best of 'em, and have shared credits with everyone from the mighty Prince Buster to the Dub Pistols.

Recorded through a well-considered chain, with premium links capturing crystal clear playing. This 1.65GB set of largely trumpets and trombones will slide right inside any tracks needing a splash of a Jamaican-flavored horn section.

So, any breakbeat bangers, jungle rollers, or dubby beats you have – run these past them to find a toot or two to make Toots proud.

loopmasters.com

TURN ON INSPIRATION
u-he
PRESSWERK
SATIN
HIVE
REPRO-5
ZEBRA
u-he.com



MIX & MASTER FOR STREAMING

Learn how the loudness rules have changed and get your tunes to make the grade online

For decades, many producers and engineers were locked in an arms race. From music's birth as a mass media product pressed to vinyl, through the CD revolution, the objective of many in the industry was to make each track as loud as possible – louder than the last, and loud enough to compete with the songs on the radio that came before and after. The loudness war escalated until 2010, when a new AES/EBU standard sought to impose a change for the quieter.

In short, today's streamed music standards leave far fewer incentives to squeeze a tune to within an inch of its life. Platforms such as YouTube, Spotify and Apple Music will actually analyze an uploaded track, turning down one that's 'coming in too hot', and turning up one that's calmer, smoother mix with more dynamics.

For listeners, the result is a far more reliable listening experience. A death metal track can be followed by a smooth jazz tune, and theoretically there'll be no need to touch the volume control – the differences in average loudness of the two songs will have been compensated for by the streaming service behind the scenes.

For engineers, this new approach to loudness means that mastering is a more forgiving process. Instead of focusing on the commercial realities of loudness, more space is available for nuance and dynamics in the music. But how do these standards actually work? And how can you implement them in your music? Read on to find out.

The document that stopped the war

You'll have heard the terms Peak and RMS when referring to audio signal level. Peak is the instantaneous maximum value of the signal – the highest point it has reached; while RMS provides a value for the average level of a signal over time. RMS gives a good representation of how loudness is experienced by humans; Peak tends to represent the maximum capacity of the system used to play it.

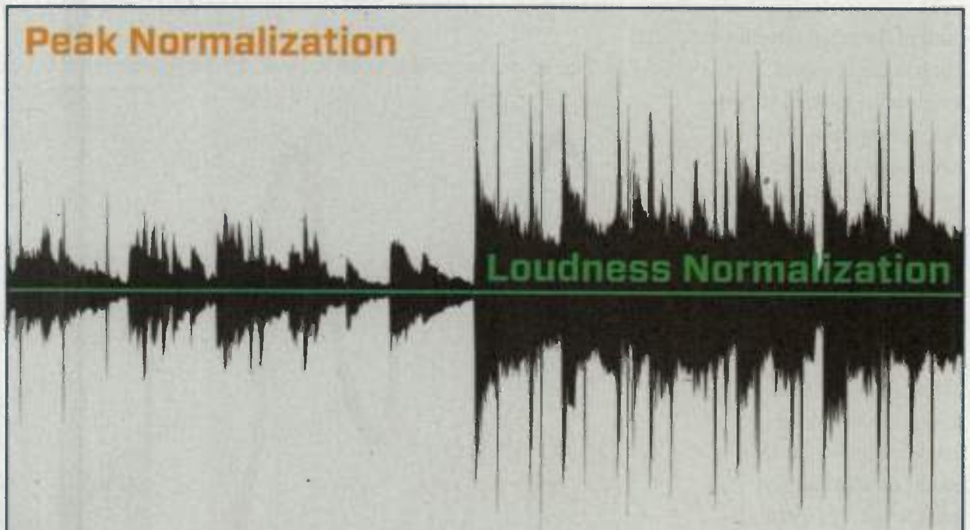
During the loudness wars, the highest peak was a set value. In the days of the CD, music was carried with an amplitude range across 16 bits. There was a definite peak level that a piece of digital music theoretically couldn't cross: when those 16 bits were full of 1s. With the Peak level set and unchangeable, the incentives were aligned for engineers to raise the RMS value as loud as they possibly could.

In 2010, the European Broadcasting Union issued EBU R 128, a recommendation for a system of Loudness Normalization. This specified a way to measure the perceived loudness of an audio signal, and crucially, a target level for that signal. If a piece of music has an average loudness above that level, it is to be turned down; if below that level, it should be turned up.

Stop! In the name of LUFS

The new system replacing RMS level (for our purposes here), is a system of Loudness Units (LU). The Full Scale value (LUFS) measures an absolute loudness and is referenced to 0dBFS. To get the actual measurement, the signal is K-weighted with a high-pass filter and a 4dB increase above 1kHz.

The EBU's recommendation is for streamed music to aim for a target level of -23 LUFS. Streaming platforms have got onboard with this somewhat, with Spotify and YouTube normalizing to -14 LUFS, Apple Music to -16 LUFS, and Amazon normalizing to between -9 and -13 LUFS.



The old ways of 'Peak Normalization' have given way to a new standard that centers music

Dynamic range: then and now

Before the R 128 standard, decades of squeezing music's dynamic range ended up leaving us very squashed. A look through the website drloudness-war.info will give you a few figures to show the extent of the situation.

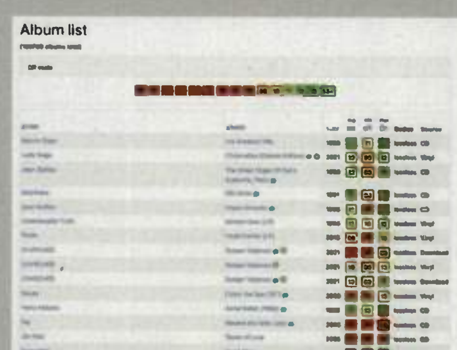
This resource lets you search for artists and albums – or to simply check out all albums at once – and to view their dynamic range stats with average, minimum and maximum figures available. While not taking into account figures such as True Peak (discussed later) and exact LU, the amount of headroom left in the tracks by the people who worked on them still sheds some light on the state of the loudness wars.

Rarely do today's albums make it close to 14dB of dynamic range, a figure the site gives as the ideal and measures with a green box. Scroll through your favorite artist's list of records, and you'll often see a sea of red. The site measures the masters of any album's vinyl, download and CD release separately, making it easy to note that vinyl releases often – although not always – offer more maximum dynamic range than their CD and download counterparts.

Another interesting thing to rank is the differences in dynamic range over time, whether that's in new material, remasters, or simply work from different artists released at different times.

Take The Rolling Stones, for example – the site shows a steadily wide dynamic range (a sea of green) throughout the '70s and '80s, although the data is often taken from CD releases of albums recorded in earlier years. Fast-forward to the mid-2000s, and the situation does start to change, with the dynamic range of the band's music being squeezed. While there are some let-ups, the general trend from then on was for the recordings to reduce in dynamic range as time went on. These days, there's plenty of variation in the aged band's recordings, but no overwhelming resolution towards being over- or under- compressed.

It's important to remember that the dynamic range statistics don't measure loudness per se, but at the very least, the website is a great way to spend some time.



dr-loudness-war.info gives you figures for the dynamic range of various album releases

Peaking too early

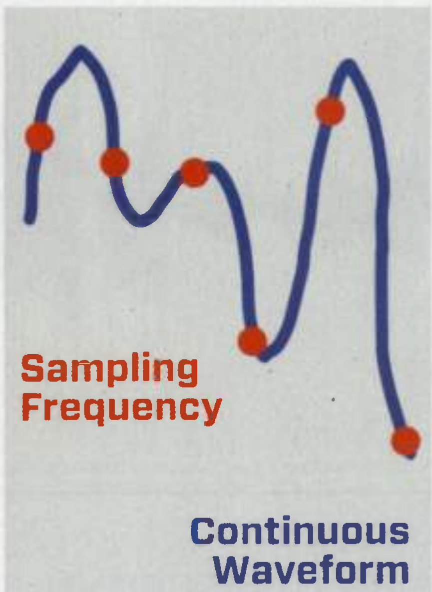
Despite the move away from using a signal's peak level as the be-all and end-all, the new loudness standards haven't ignored the concept. True Peak is a new way to measure the loudest points in the audio signal by recreating how a real, continuous audio signal actually behaves. MeterPlugs' Ian Kerr explains for us...

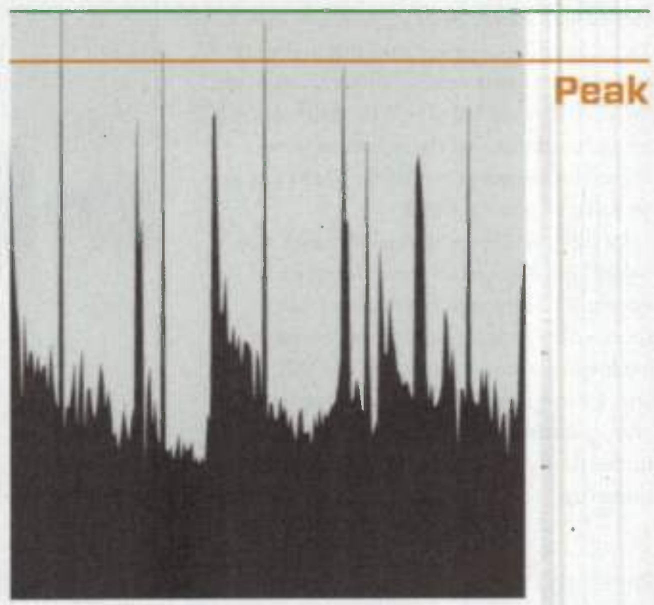
"A typical 'sample peak' meter will simply look at discrete samples to determine the peak level. The problem with this is that the samples may not have been taken when the continuous signal was at its peak, so in these cases the sample peak meter will report a lower peak level than the actual peak level.

"This can lead to problems further downstream when processing decisions are based on this 'faulty' peak level. For example, suppose a mastering engineer applies compression and limiting to track, aiming for a sample peak level of -1 dB. It's entirely possible that the actual, continuous peak level will exceed 0 dB in this case! When the track is ultimately converted back to a continuous signal for listening (by a Digital to Analog Converter), this could lead to distortion. It can also lead to artifacts when the track is encoded.

"True Peak meters aim to mitigate this issue by detecting inter-sample peaks: the peaks that occur between discrete samples. They do this by upsampling the digital signal, from 48 kHz to 192 kHz, for example. This reduces the likelihood that a 'true' peak will be missed, or reduces the amount by which it is missed, at least. The result is that True Peak measurements more accurately reflect the actual peak levels than a sample-peak measurement."

The True Peak standard aims to account for missed overshoots by oversampling the signal





Loudness range issues

Loudness may be measurable, but over what time span we do it can make a difference. The EBU Mode specifies three sliding time windows over which to measure loudness. Momentary has a window of 400ms, which helps to meter transient and highly dynamic material; Short-term increases the window to 3s, giving a wider view of the loudness in a musical context; Integrated keeps track throughout an entire track, and can shed some light on the potential pitfalls of the new methods for working with loudness.

Stop! In the name of LUFS

If LUFS and True Peak are taking the helm when it comes to measuring loudness, and are responsible for streaming services' fiddling with the volume of any track on playback, then how do we account for the fact that music changes in loudness throughout individual tracks?

This is measured with LRA, a Loudness Range number that describes the integrated difference in LU throughout a track. Let's say a piece of music starts with some ambient sounds and background noise, has a light, acoustic guitar intro and eventually builds to a momentous climax before bringing things right back down again – the LRA for this track would be relatively high. On the other hand, a modern techno tune without much variation may remain quite consistent across its length, which will then give us a lower LRA reading.

It seems streaming services are not currently responding to LRA when calibrating uploaded tracks, so even if a relatively small portion of the audio signal gets particularly loud, the entire track will be compensated for based on this one portion. Even the quieter parts will be turned down.

EBU R 128 may have established a steady peace after the loudness wars, but one unintended consequence is that it incentivizes a reduction in the LRA throughout any given track, and so entire pieces of music may end up with fewer changes in loudness per section.

Step by step 1. Using Mastering The Mix Levels to prep a track for streaming

1 Let's put some of this mixing and mastering knowledge into practice. Here's Mastering The Mix's plugin Levels, which lets you know how your audio is shaping up against various commercial and theoretical standards. Levels goes on the master buss, after all processing.

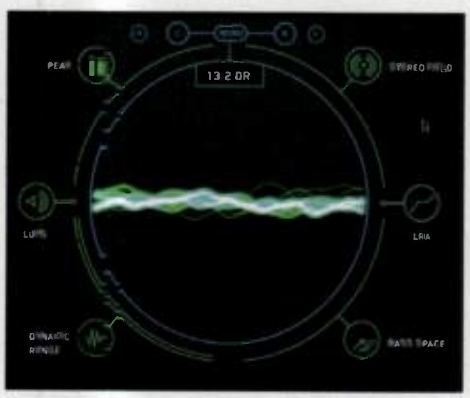
2 As we play our music back for the first time, we can see from the icons around the circle whether our track is shaping up at different times, or whether there are some issues. We'll head first for the Peak detector, which shows us that the mix is peaking at a nice, safe -1.0dB... or is it?

3 By switching to True Peak mode in the Settings screen, we can see that the signal is really peaking at -0.6 True Peak, which would usually cause it to be turned down until it reaches the desired -1.0. We can use this knowledge to reduce our transient levels, while leaving other, average-loudness-causing elements intact.

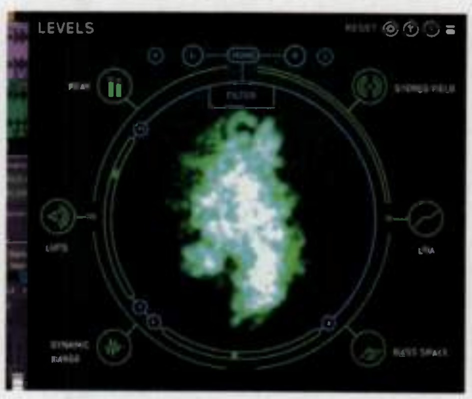
4 Now onto the other crucial part of this article: the LUFS meter. Our track's Short reading (about a 3-second window) doesn't tickle the meter, hovering between -15 and -13 LUFS. While there may not be much green here this sort of reading is actually ideal, and would lead the track to sail through any streaming checks quite unharmed.

5 The Integrated reading for Levels' LUFS metering gives you an idea of the LUFS reading throughout the entire track. This can help get a decent reading, especially if the material from the start to the finish of the track varies heavily in its loudness.

6 The LRA of a track tells you the difference in loudness between its loudest and quietest parts. This is something to keep in mind, knowing that your track's loudest part will likely be the one that's calibrated to the target loudness of each streaming service, and the quieter parts could end up too quiet.



7 Dynamic range may not seem to be as important as it was now that we're in a world of LUFS, but the same guidelines as ever should apply: squeeze your track too much, regardless of the measure loudness, and transients will lose clarity.

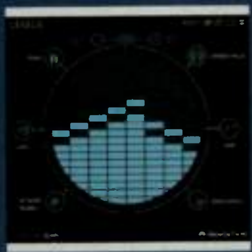


8 Although the final two processors of Levels aren't necessarily anything to do with loudness as measured, they can still help at this stage. Stereo Field measures width, and phase coherence, and lets you apply low and high filters to mono the tops or bottoms of your track.

9 Bass Space, meanwhile, gives some insight on whether there higher track elements leave enough room for the kick and bass. You mute these two elements before doing the analysis. This knocks our Stereo Field reading out for the moment, but it'll come back with the full mix.

MASTERCLASS

THREE OF THE BEST LOUDNESS PLUGINS



MASTERING THE MIX

Levels

$69

A simple way to gauge whether your mix or master is hitting various commercial targets or not. Levels puts several properties around its central display, and turns each icon green or red, depending on how it's going. If you're not measuring up in any area, you can get more info on why in the central readout.

masteringthemix.com

YOULEAN

YouLean Loudness Meter 2

FREE / $47

One of the first great solutions to interrogating your mix as to its LU and True Peak, as well as many other properties. You can get the free version of Youlean's Loudness Meter, or plump for the pro, which won't break the bank.

youlean.co

METERPLUGS

Loudness Penalty

$49

Get a semi-instant recommendation on how much the different streaming services will turn down your track. Loudness Penalty analyzes your current work and also keeps track of changing standards, helping you avoid surprises.

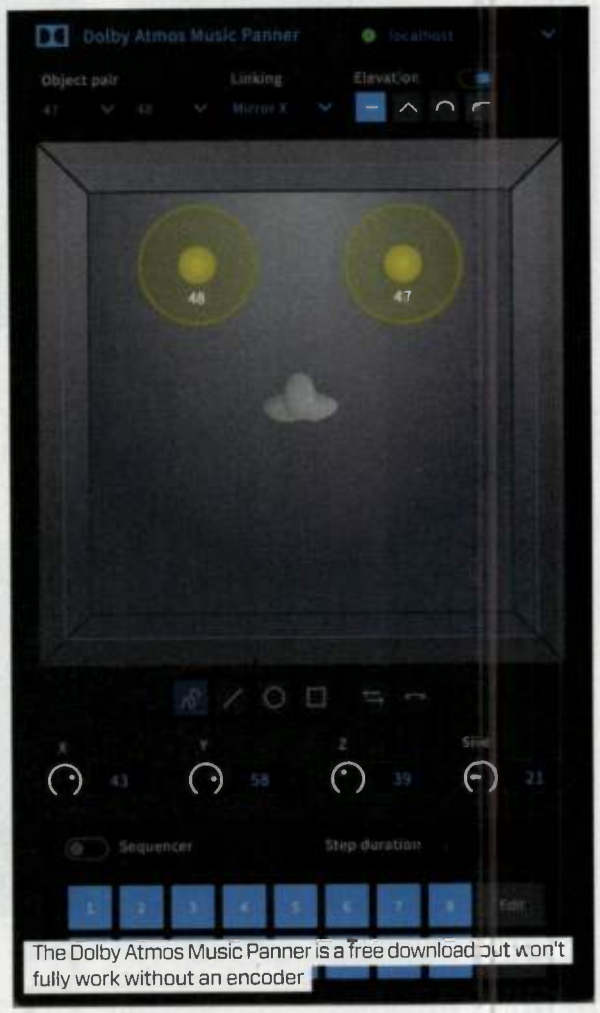
meterplugs.com

Mixing in Dolby Atmos

We understand the world of stereo: the audio signal consists of two channels, and then differences in level between one signal appearing in both channels create the impression of the sound appearing from somewhere between the two speakers. Those levels can be changed and adapted, and a sonic object can be made to appear in any place along the line, or indeed move around it.

Dolby Atmos take this further. The concept is simple enough: instead of having just two speakers as in a stereo setup, or five speakers and a sub as in a 5.1 surround sound system, Dolby Atmos allows you to play back over whatever system you do have, and will decode the same signal differently, but with compatibility, which means that you can play a full-range Dolby Atmos mix both over a theatre setup or simply by using those two speakers in a home stereo formulation.

A Dolby Atmos setup is denoted by three numbers. While a classic surround setup may be called, say, "7.1", standing for seven speakers and a subwoofer, one example of a DA setup could be "7.1.2", allowing for two speakers placed as a stereo pair above the listener. So how does compatibility work with the system? And what's going on behind it?



The Dolby Atmos Music Panner is a free download but won't fully work without an encoder

How it works

Because Dolby Atmos will play back based on whatever speaker formulation the listener is using, the way it's handled by a mixing engineer must be all-encompassing, allowing room for translation. The way it's done is by mixing in a 360° 'dome' around the listener.

As shown in the tutorial on the following page, panning is done using the Dolby Atmos Music Panner plugin, which allows sounds to be placed anywhere in a 360° circle, and to allow for their elevation as well. With the Panner plugin finished with over a full project, the positional values of each instance can be encoded, and then on playback an opposite system will decode these into audio signals that meaningfully convey the position on whatever the listener's system is using.

As for the console (or, more likely, the virtual-console) workflow, a typical Dolby Atmos project will have 64 channels, although 128 are possible. A 'Bed' is an output channel that corresponds to a single speaker in a particular Dolby Atmos system that the engineer is using to play the project back. In our tutorial project, these Beds take up ten channels: seven for the head-level speakers, one LFE channel, and two top-front speakers above the engineer.

The remaining 54 channels are taken up by 'Objects'. Each of these Objects is a pair of channels that can be given its own positional information. Using the Dolby Atmos Music Panner plugin, you can choose positional locations (or movements) for each of these different Objects, and their signals should, when encoded, then reach the Beds in the correct amounts.

Step by step 2. Working with Dolby Atmos Music Panner

1 Let's have a look at how Dolby Atmos can be encoded in a DAW using the Dolby Atmos Music Panner plugin. The plugin is available in Audio Units, VST3 and AAX versions, and comes for free from Dolby's website. At the same time, you can install various template projects for a few of the major DAWs.

2 Here in Ableton Live, the template project loads up with ten mono tracks and 27 stereo ones (ie, 54 mono pairs). A copy of the Dolby Atmos Music Panner (VST) is loaded onto each of them, routed and ready to go.

3 The idea is that the ten mono tracks (the Beds) represent your system's outputs, while the remaining dual-channel pairs (the Objects) are where the setup work happens. You should adjust the Beds setup based on your playback system, and output each of its channels to the correct place.

4 Now to start using the Panner plugin. We add one of our mix elements onto one of the Object channels, 19-20. Opening the plugin on that pair, we can start moving the '19' and '20' pucks around the virtual listener's head in the center of the square. With a renderer connected, this will translate the positional information into the correct strength of signals.

5 Because this pair is in Mirror X linking mode, moving one puck will move the other in an equal and opposite direction. In other words, the two are mirrored as if there's a vertical line in the center of the square. The pucks also show green and yellow in order to provide a form of metering for the audio coming in.

6 Alternatively, setting Linking to Mirror Y does the same thing as if a mirror is placed horizontally in the centre of the square (as pictured). Mirror XY mode does both, allowing the two members of the pair to be at opposing diagonals, front and back, or left and right... or anything in between.

7 We can alter an object's Z position using the knob below, bringing it further towards the top of the playback environment. This would only work on a system with top speakers, of course. The Size control aims to make the virtual source appear larger, bringing on more speakers but still giving precedence to whatever direction is selected.

8 Using the Elevation selector, you can choose a ceiling profile for the environment. By default, it's a flat line, and only your Z position tweaks matter. With a pitched or domed elevation selected, the object moves upward as different rates based on its difference from the center.

9 At the bottom, the Sequencer can be activated to make impressive, moving effects. By clicking Edit, you can prime the interface to program movement shapes using the tools provided below the square. There's a lot to do here, although it may be more for movie sound than for tasteful mixing practices!

MULTI-EFFECTS

These plugins offer most of the effects you could possibly need in one single place, but which ones are worth the investment?

HY-Plugins HY-MBMFX2 $48

Flaunting 22 effects and multiband processing, HY-Plugins' effects unit lays down the marker in its field. Virtually endless possibilities stem from the effects which include delays, filters, modulation effects, distortion and reverb. These can all be chained in any order in each of the three frequency bands.

LFOs, envelope followers and macros can be routed to trigger the effects' parameters. You get two slots for each parameter, vastly extending the control and movement of the plugin. The three frequency bands then feed into a master effects and a three-band EQ.

The ability to reorder modules blurs the lines between multieffects processor and channel strip. You can use it for anything from subtly applying overdrive and modulation to vocals – or exploring the expansive routing options to build deep, moving soundscapes.

hy-plugins.com

AudioFB Neutrox 2 **$39**

Neutrox's four modules appear to align with the "two sides of the brain" theory. The left side of this multieffects unit promotes methodical time-effect tweaking. The reverb module has two tweakable environments, a function-heavy delay module offering a variety of beat divisions, and an inventive Width parameter. The right side is more creative: a pitch-shifting resonance filter and an LFO saturator with a Lo-Fi control provide a wacky modulation dreamland.

Neutrox provides characterful modules each with a dry/wet blend. The plugin is best suited to an effects bus where blending extreme modulation with the dry signal adds dimension. This approach maximizes the plugin's functionality while limiting abrasiveness. Use sparingly to add reverb, delay or subtle modulation – or to coat a sound in a thick layer of flavorsome juice. A useful tool.

audiofb.com

UJAM Finisher FLUXX **$39**

Following the latest trend of algorithmic multieffects plugins, FLUXX brings an entire rack's worth of parameters to five interchangeable dials. Offering a simple, inspiring interface, users can trigger their imagination and lift dull sounds. The dials change function with each preset. FLUXX has four modes; Chords, Arpeggio, Rhythm and Solo, categorizing the presets and helping the user narrow down which to choose. Our honest opinion is that unless the user is familiar with all of FLUXX's presets, algorithmic tools may hinder artistic choice in this case: FLUXX has a steep learning curve. The "Finisher" doesn't quite live up to its name and makes the signal muddier than to begin with, and we find ourselves putting FLUXX at the start of a signal chain rather than the end.

ujam.com

Cableguys ShaperBox 2 **$268**

ShaperBox lets you draw custom curves to influence its four components: TimeShaper, the new DriveShaper, FilterShaper, CrushShaper, PanShaper, VolumeShaper and WidthShaper. You can, for one example, use VolumeShaper to create an LFO-style signal that brings the level of your track up and down rhythmically, in a shape you draw yourself. Change the timing of the curve (ie, LFO Rate), draw custom patterns and create steps, process different curves for Low, Mid and High parts of the signal, and go deep with editing. It's a deep experience, but thanks to preset curve shapes, MIDI control, presets for each 'Shaper, it remains intuitive and creative. The one drawback: it doesn't offer quite as wide an effects range as you'd get elsewhere.

cableguys.com

NI Guitar Rig 6 $246

Also available as part of the huge Komplete bundle, Guitar Rig 6 is a slick, multi-faceted amp and effects modeling software package. Aside from being a reliable tool for creating realistic guitar tones, GR6 has a bristling armory of effects that will completely transform whatever audio they're employed on. Compressors and reverbs modeled on vintage studio rack modules, characterful modulation effects, mixbus plugins and much more; it really is difficult to fault NI for the sheer variety this software possesses.

The updated interface is easy to navigate, with a side panel that will clearly direct even the least perceptive among us to presets you can fit to your input source, or components for building an original rack. Guitar Rig 6 is a really useful producing tool – use it to beef up frail effects, or an all-in-one processor for any instrument.

native-instruments.com

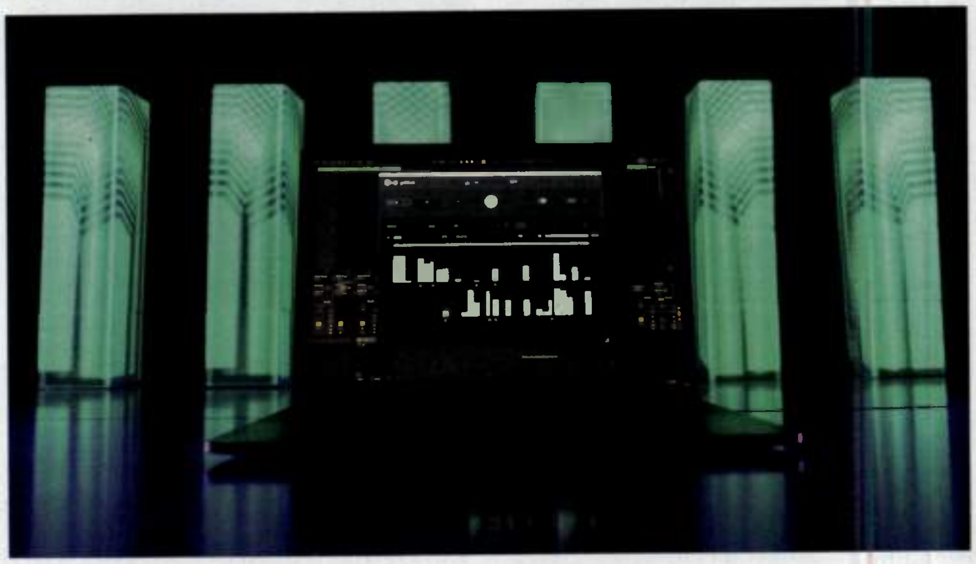
I Got Rhythm

Gatelab offers a new route to creative comping.

By Michael Ross

Every guitarist knows about tremolo (though they may mistakenly call it vibrato). Found on guitar amplifiers from their earliest days, as well as in numerous pedals, tremolo is the rhythmic raising and lowering of an electric guitar's volume. In its classic form it is an effect that applies a moody, vintage vibe to plucked low strings or strummed minor chords — think Ennio Morricone.

But rhythmically affecting a guitar volume signal has the potential to produce many more effects than just the spaghetti western sound we know and love. Audiomodern's Gatelab, available as a free plugin for Mac or Windows and as an app for iOS, offers myriad musical ways to modify your guitar signal volume through sequenced gating.



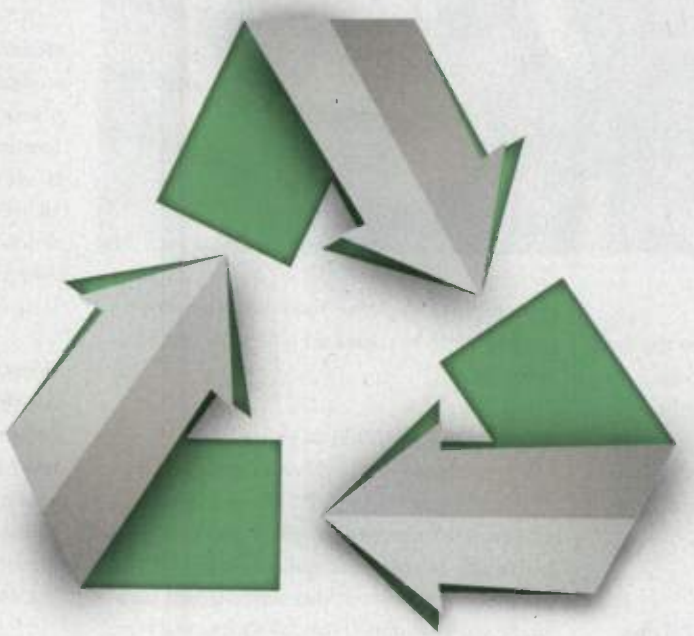
When you open the app in your DAW or iOS device, you will see a pair of grids and a plethora of ways to modify their parameters. There will be a number of graphic bars on the grid representing the volume of your signal at that point in a two-bar musical phrase. When you click the big button at the top, the location and size of the graphical bars will change creating a new rhythmic volume pattern. It opens in stereo with two grids, but you can switch it to one grid for mono. You can link the two grids to have the same pattern emerging from each of your speakers or unlink them for a cool, random ping-pong effect.

The app works in two modes, Flow and Gate. In Gate, the volume of each step in the sequence is either on or off. With Flow mode, the volume can vary anywhere between full on and full off. You can set the sequence to run forward, backward or ping-pong between the two. Pushing the button will randomly change the pattern, but you can pick one of three levels of density, which will remain constant, from only a few audible steps to only a few silences. You can also lock a step so it will remain when the others are randomized. This is useful if, say, you always want to have audio on the first beat. Gatelab can also be set to continually, automatically randomize the pattern after a chosen number of bars, from one to 64, without pushing the button.

You can draw in your own patterns of gating or flow, choose the timing relative to the BPM, from 1/8 to 1/128 (with various triplet modes), and pick your preferred number of steps. These steps can be subdivided, faded in and out, and more. As a bonus, Gatelab can also be used to control the MIDI parameters of other effects in your DAW.

Given the possibilities, it is great that the app includes a preset function. There are plenty of included presets or, when you find a pattern you prefer, it is very simple to save it: click one of the 16 preset slots on the GUI and done! A preset management page lets you name them, move them around, or trash them.

Using my guitar through a fuzz, I was able to quickly set up a Moog-like sequence, drop it into an Ableton clip, and lower it an octave for a terrific bass part, locked to my kick drum. For creating a wide range of evolving rhythmic effects from pulsing, chill pads to strident EDM sequences, Audiomodern's Gatelab turned out to be an inspiration machine. Did I mention it is free? ■



CID RIM

Austrian producer and multi-instrumentalist CID RIM injects the fluid energy of live performance into his jazz-inflected electronica, putting his skills as a pianist and drummer to work in genre-spanning productions for the Scottish label LuckyMe. We caught up with the producer in advance of his latest record, *Songs for Vienna*...

When did you start making music?
I started playing piano when I was six. Playing mostly classical music until about 11, I then found out I can just play whatever I want, without the notes, very liberating. I started trying to find out the chords of the records I was listening to. Like the Rhodes loop of Janet Jackson and Q-tip's Got 'Til It's Gone. Later I started playing drums, being in bands and at about 14, a friend got a copy of Reason 1. The possibilities of having a DAW with just a simple synth, a sampler and a drum computer basically blew my mind. I was using my headphones as a microphone, sampling jazz records, recording all sorts of everyday percussion objects, like playing using a glass and a pencil as a ride cymbal.

Tell us about your studio/setup?
I use a rented studio in East London, it's not big but I'd say the most important thing about it is a window with daylight shining in. My studio is part of a bigger studio complex with a bunch of rehearsal rooms, so I can just walk in one whenever it's free and play drums, which is brilliant too.

I have great ADAM speakers that came with the space, two monophonic and one polyphonic synth. A MIDI keyboard and a decent Neumann vocal mic. That's all I need really. When I finish up records I often go to a mate's studios and use their fancy old analog gear. And when I get a good old synth into my hands I record right away, and see what happens. Often those initial get-to-know-a-synth moments are where the best ideas come out of.

What DAW (or DAWs) do you use?
I still mainly use Reason. I went through all the different stages, starting off with no audio or plugins, just their onboard effects and MIDI. You had to play any recorded audio through the sampler, a very limited process that taught me a lot. I'd compare it with footballers learning how to play with a tennis ball on a concrete pitch. But I wouldn't say that was a disadvantage at all.

I also use Ableton for recording multiple tracks, like a full band or drums, and for playing live too. Sometimes I make a track in Ableton, but I still prefer Reason. Seeing the cables bounce when you press tab just makes me happy somehow.

What one piece of gear in your studio could you not do without, and why?
I need my laptop, my £27 Sennheiser in-ear-plugs and a MIDI keyboard. The first two of those are always on me and the MIDI keyboard is the easiest thing to get everywhere. Even the laptop keyboard is fine. I love making music while traveling, on a train, in a hotel room, in the pub, anywhere really.

If I count all the spaces I've worked in on this album I'd say that's been about 15 different studios, and probably another 30 weird random places. I'm quite flexible, I also don't keep synths forever, one must go when another one comes in. Never more than three at any one time and they all need to be tiny and light. I guess with playing drums for all my live performances there comes a deep, true, genuine hate of carrying heavy things. [*Laughs*]

What was your last purchase?
An AKG C520 L headset mic for playing drums and singing at the same time. Technically that's more a piece of live equipment, so the latest studio gear I've bought is the Roland Jupiter Xm. I still need to properly dig into it, but simply playing around with a regular triangle wave and the filters already sounds amazing.

What dream bit of gear would you love?
If I had to pick one thing it would be a grand piano. Just last week I recorded in a beautiful old, huge radio studio in Vienna, Studio 2 at Funkhaus. It's where they record entire orchestras for radio.

In there they have two Bösendorfer grand pianos, and one of them has the lightest touch to the keyboard I've ever felt. So one of those please, thank you!

ESSENTIAL TIPS

You're always right
Intrinsically a paradox, but I mean it: when someone tells you how a certain thing should be done, then you should get wary, in fact maybe try out how the exact opposite sounds first. Only you can create the music you're making, therefore I believe you're always right.

Train your ears
They're all you've got really. No ears, no music. The better you hear, the better you play, produce and mix. Go into that micro timing and start to feel the tiniest 5ms difference. Play, sing, record, learn every instrument. Listen closely, make a decision and know the frequency that you want to get rid of before you're even opening the EQ. Make decisions quickly, don't get lost in details. The best live tech person I know was an incredible child prodigy violinist. Being interested in playing instruments and singing, maybe even a bit of traditional ear training, will help you.

Sidechain certain frequencies
I love that FabFilter Q3 option where you send another track into the EQ and that track triggers your dynamic EQ curve. It'll even show you where the two tracks are clashing on the frequency spectrum in red, so just get rid of the frequencies where you see a lot of red.


One player's unused gear
is another player's grail.
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and make room for something new.
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