DIGITAL FISH PHONES DOMINION

Plug-ins, as a concept, are brilliant, but creating the language of ones and zeros needed to simulate the subtle and endearing qualities of a classic equaliser or compressor is far more difficult than getting the graphic interface looking like the original article.

I used to argue that you should minimse the number of transitions from analogue to digital and digital back to analogue as possible due to the degrading effects of every A/D and D/A converter. Having tried a few plug-in compressors, my views have (temporarily, hopefully) changed. The destructive nature of even the most basic D/A A/D converter pales in comparison to the destructive potential of most plug-in compressors, and therefore your final product will sound more expensive if you go out of the digital box for some quality analogue compression, if needed.

Digital processing involving EQ and levels is adequate but it seems that almost any processing that involves elastic dynamic timing is horribly wrong, stiff, crunchy, noisy, jerky, stepped – anything but fluid and musical. Tasteless really. For that reason it's worth stepping through the digital to analogue doorway long enough to bounce around inside a real compressor. Among other things, this gives you back fluid movement with infinite resolution in the level changing department.

Having said that, the sheer flexibility of digital processing has produced a few interesting plug-ins not possible in the analogue world. One of which was written by a codesmith by the name of Sascha Eversmeier, from Berlin, Germany.

Dominion version 1.2 is a dynamics controller. Rather than reacting to the sound level and attacking or releasing with a given ratio, Dominion looks at the shape of the sound's envelope to detect where the head and tail are and then proceeds to hand you a slider to control the relative level of these two components. It's almost freaky the way you can raise or lower the transient of any sound by dramatic amounts without affecting the rest of the sound. This is a free VST plug-in which is part of the digital fishphones family of plug-ins. (www.digitalfishphones.com). If your mouse doesn't seem to let you grab the slider, try right clicking and dragging.

I've yet to find a pleasantly expressive distortion plug-in that simulates anything analogue.

Stav

MY FAVOURITE PLUG-IN

AT writers and industry pros reveal their secret sonic software weapons.

DUY DAD TAPE & MCDSP ANALOG CHANNEL

I've always had a soft spot for valve and tape emulation plug-ins. My favourites during the OS9 era for Macs was the Spanish firm DUY's tape emulation. DaD Tape got a right workout as I'd apply it to every single track in a session. It got to the point where I wouldn't even listen to the tracks, I'd just render each track offline with the Audiosuite version regardless. Just recently DaD Tape has become available for native systems in RTAS, AU, VST and MAS format. It's really great for giving everything that extra bit of wallop. And while the change is minor when you listen to single tracks, it really makes a difference on the entire mix.

Since moving to OSX, I had to find a similar plug-in, as at the time, the DUY plugs weren't available for the OSX platform. About the only thing on offer was McDSP's Analog Channel, which includes two TDM plug-ins: AC1, which is a console channel strip emulator; and AC2, which does the tape emulation thing. While I didn't like AC2 to begin with (it didn't seem to offer the smoothness of the DUY version), I've since come to fall in love with it, as it's much more configurable. I'll still render most tracks with a single setting I decide on using Audiosuite processing, but then I find it's great to use the bias, roll-off and 'bump' settings to tailor the sound further – as if using those parameters as EQ in their own right. I love it on pretty much everything.

Brad Watts

"It got to the point where I wouldn't even listen to the tracks, I'd just render each track offline with the Audiosuite version regardless."

LINE6 AMPFARM & SERATO PITCH 'N' TIME

Hmm… for an analogue guy, I'm enjoying digital very much these days. Most of my recording is done at 24-bit/88.2k, on a ProTools system rigged with either Apogees or the Digidesign 192 converters. For all the tech heads who decry the Digi converters, they've served me well – all the Led Zep and Iron Maiden projects, as well as Metallica, Page/Plant surround mixes (and everything else I've done in the last five years), have suffered quite well being 'scrambled' by a 192!

However, I do not mix internally at all – I just love the art of moulding sonics in the analogue domain, so I use the ProTools rig as my 'tape machine', feeding individual tracks into an SSL 'K' series console. Anyway, my approach to plug-ins is as follows:

I do use the delays and sometimes apply [Line6] AmpFarm to crush a bass guitar or for a vocal effect. I find the MetaFlanger easier to manipulate than the analogue equivalents, but when it comes to compressors and EQ, I'm afraid to say, analogue still wins hands down, and if there's an 1176 in the room, chances are something will be screaming into it!

Serato's Pitch 'n' Time has been essential for me also, in massaging multi-track formats recorded with no timecode, to fit unsynched video.

I see the day approaching when plug-ins will dominate my audio landscape. But, that said, I've just purchased a Studer A827 analogue 24-track recorder last week – because *nothing* sounds like analogue yet!

Happy recording. And remember, nothing works better than a good microphone, correctly positioned, to capture the nuances of a fine sounding instrument or sound.

Kevin 'Caveman' Shirley
www.CavemanProductions.com

> "I'd love to see all the plug-in manufacturers get with the 21st century … and start building plug-in windows that can be *resized*"

WAVES SSL 4000 E-CHANNEL

Picking a 'favourite' plug-in from the hundreds I've used lately is a difficult task. If I had to nominate one right now it would probably be the Waves SSL 4000 E-Channel. I've been using it for all sorts of tasks during mixdown, from vocals to guitars, drum submixes etc, and it works incredibly well. Apart from having a conventional SSL console parametric interface (ie., no rollercoaster waveforms), and a built-in comprehensive dynamics section that I can easily relate to, it's the *sound* of the plug-in that I find most appealing.

You can use the EQ on just about anything, even as a sidechain for the compressor to help de-ess vocals. The expander/gate works a treat to dry out boomy and roomy drum mics and the filters – as I've often stated in the past – are extremely powerful tools for controlling the bandwidth of a sound. SSL channel strip filters are powerful tools on an E-Series console, helping hugely to control clarity and focus in a mix; there's nothing worse than hearing 10 elements in a song that could have done with a bit of filtering, if only the mix engineer had thought to engage them. Probably the only thing I find a bit annoying about the Waves SSL E-Series Channel – which is also the case with channel compressors on a real console – is the fairly basic gain reduction metering on the compressor and gate: one 'LED' for every 3dB of gain reduction in the initial stages… which might have been a limitation of the physical layout of the board once upon a time, but something that could have been easily modified in the plug-in.

There are a couple of other plug-ins that I feel compelled to mention here. One is the Eventide reverb, which sounds superb to my ears and is very easy to use… I particularly like the ability of the plug to filter the reverb and delays as well as mix the delays and reverbs into one another – a great reverb plug-in that I use all the time now. The other one is the Universal Audio's 1176LN plug-in. Sounds great for a digital compressor and can be used for all kinds of compression tasks…

As a closing comment I'd love to see all the plug-in manufacturers get with the 21st century sooner rather than later and start building plug-in windows that can be *resized*. I have a 24-inch Dell screen and half the time, when I'm using plug-ins, I still find the parameters within them impossible to read. If unlimited resizeability is too hard initially, just give me three presets that I can click on: small, medium and full screen, rather than lame excuses about why it's 'impossible'. Unlike the analogue domain, where at times you're up close to, say, a compressor, while you set up its gain structure, and then later you might only be glancing at it out of the corner of your eye, with a plug-in it's one-size-fits-all irrespective of whether you're staring at it to the exclusion of everything around it or not – which sucks the big Kahuna. Once someone offers this feature, everyone will fall into line and we'll all be wondering how we ever put up with this crap for so long.

Andy Stewart

FREE PLUGS

I perform live improvised electronic music. Often I'll extend the AudioMulch (www.audiomulch.com) environment with VST plug-ins to give me access to effects that aren't natively available in the program. I tend to prefer free or affordable software over high-end studio-oriented plugs, since often the sonic quality differences aren't noticeable in the kind of bar/club venues I usually play in. I also believe that finding the most musical ways to use what you have can be more important than having the best gear. My most oft-used plug-ins at the moment are a combination of studio-staple processing and wacky out-there stuff, here's a few of my faves:

Classic Series plug-in pack by Kjaerhus Audio (www.kjaerhusaudio.com/classic-series.php) is a freeware suite of basic studio effects. I mainly use the compressor and mastering limiter for live performances where I want to keep some or all of my mix under control, even if I do something stupid.

Then there's Ambience by Magnus at smartelectronix (magnus.smartelectronix.com) which is a shareware reverb that has all the configurable parameters you'd expect and sounds pretty good to me. Two features I really like about it are the 'Hold' button, which lets you freeze the sound into a drone, and the 'Quality/CPU' knob, which lets you trade off CPU usage against sound quality, which I find handy when my patches start maxing out my PC.

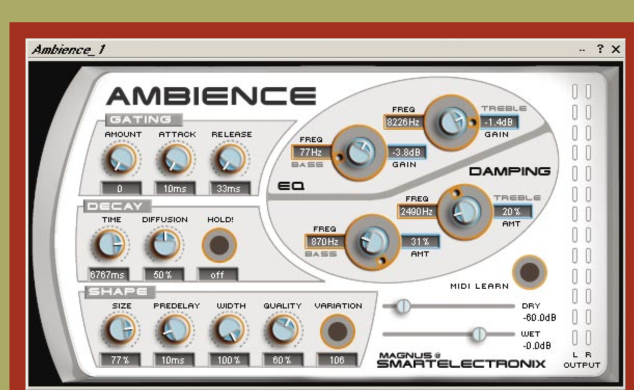
Another one of my favourite freebies is SIR – Super Impulse Reverb, by Christian Knufinke (www.knufinke.de/sir/). This is an efficient plug-in for use on arbitrary sample as a reverb 'impulse response'. But I don't use it for reverb. I more often use it with non-reverb sounds, like recordings of plucked or bowed string instruments and then play granular sounds through it, which gives you a kind of filter where SIR acts like a sampler, playing back samples filtered by the spectrum of whatever you feed into it. Feeding in dense noise or granular textures is a great way to make rich drones. You need to experiment with what samples to use in it, and what sounds to feed through it, but it gives a lot of useful possibilities.

Beatfckr VST is a great old (hard to find) plug-in, that is capable of creating some great varied, random, IDM-type glitchy beats. Beatfckr provides a chain of delay/ringmod/bitcrush/delay effects, which get switched in and out and randomised whenever the input signal crosses a threshold. You can set it up to make random effects on each different beat (or only on the loud beats for example). I love it.

I know I'm probably pushing my luck trying to mention a couple more but my second last fav is LiveCut by MDSP at Smartelectronix (mdsp.smartelectronix.com/2005/07/livecut.php) – another great plug-in from the smartelectronix collective. This one provides a number of processes for automatically cutting and splicing beats in a kind of break-beat fashion. This is actually a reimplementation of some beat-slicing algorithms from Nick Collins' BBCut system for Supercollider. The algorithms are modelled after the music of artists like Squarepusher and Aphex Twin. There are a few different modes to choose from and lots of parameters to play with. You can feed in any basic beat and get something interesting out the other end.

Finally, while writing this I came across George Yohng's W1 mastering limiter (www.yohng.com/w1limit.html), an accurate clone of the Waves L1 limiter. It's freeware and looks set to become a new favourite for me.

Ross Bencina
Electronic musician, creator of AudioMulch



WAVES C4

My favourite work tool for processing is the Waves C4 multiband compressor/EQ. It is hungry for processing power, but uses it to great effect, providing wonderfully transparent control over dynamics with the added benefit of being able to individually set thresholds, ratios etc within four frequency bands that are also adjustable. Great for vocal and overall mixes and to use when subbing and bouncing tracks to give you a degree of control without too much colour.

My other favourite toy is the Tape Delay in Logic, nothing special, but it always works and sounds cool, and I guess that alone makes it special.

But if I had a hammer it would be the C4.

David Claringbold
Technical Director, Sydney Opera House

WAVES 16-BIT DITHER

Waves 16-bit Dither (OS9). Stick it on your master fader. No worries.

James Wilkinson

URS CONSOLE COMPRESSORS & LEXICON PSP 84

URS's three console models (based on 'era' and labelled by year) are among my favourite compressors. These three compressors sound unbelievable. They really beef up signal and alter the dynamics as well as the 'sound' in a very natural and appealing way. They are all subtle, however, they each produce three very distinctive sounds. Lately, I've been using the 1975 console compressor on bass, in particular, as I find it levels the bass tone very evenly. Compressing at high ratios is not an issue with these compressors either; they really squash the signal well before they start affecting the source material negatively.

Lexicon PSP 84 – A delay like no other. There are two independent delay lines, each with variable sampling, adjustable tape saturation emulation, resonant filters, wet/dry equalisation, high cut-off filters and five LFO waveforms. It'd be the most comprehensive delay plug-in I've ever seen and it sounds absolutely killer. For years I've been using the PSP 84 more as an 'instrument' than an effect. I like chopping up drums, guitars, and keys (frankly, almost anything) and applying large amounts of the 84 while flicking through the presets. The delay sounds themselves are so dynamic and varied that you can easily end up with different compositions based on what the delay is doing to your source material. I've also used the 84 as an insert on a track and with the right settings I can produce very lush moving pad sounds that often fit perfectly in the mix and instantly inject atmosphere.

Anthony Touma

OPEN SOURCE FREEWARE

I don't have a favourite plug-in… I try not to get attached to technology but aim to get a job done with whatever happens to be available. Hardware, virtual or mixed studios can be equally inspiring places to create.

That said, I welcome any opportunity to work with Waves mastering and audio restoration tools. There's nothing like them for high quality, precise, analytical audio manipulation, yet they feel (and sound) like classic, high-grade hardware.

And I couldn't be without the new plug-in version of Celemony's Melodyne. It's primo for basic pitch correction and indispensable for stretching audio that little bit further.

I'd also like to vote for a favourite plug-in that isn't: the M-Class mastering suite of compressor, EQ, stereo image and audio maximizer supplied with Propellerhead's Reason really should be spun off as a stand-alone product so I don't have to keep faffing around moving big audio files in and out of Reason for processing.

Day-to-day, though, I use lots of freebies. This is a conscious decision: Open Source rules for me. To follow in my footsteps, just visit: www.espace-cubase.org, www.freesoundeditor.com or the classic www.kvraudio.com. With the first two, you'll need to navigate to the English bits before discovering vast collated lists of free VST plug-ins – effects and instruments, PC and Mac – from all over the place. There's some dross out there but there's also a lot of very clever programming happening outside commercial R&D.

Looking at my average session, I find a couple of plugs are regularly used. For example, I always seem to use PSP's PianoVerb (www.pspaudioware.com) and Silverspike's Room Machine RM844 (www.silverspike.com). They've both been around – in plug-in terms – for ages, but they're simple, don't overload a CPU, and they sound great. Think about what happens in a piano when you press and hold the sustain pedal and yell onto the strings. This is sort of the idea behind PianoVerb. It adds subtle but tuneable ambience to sound that is like no other reverb. RM844 similarly steers clear of huge rooms and large delays and provides just what's needed to place sounds in a space if you like to mix dry, but not too dry.

To boost your plug numbers in one go, Maxim Digital Audio's suite is and essential download (http://mda.smartelectronix.com). The collection covers delays, multi-band compression and distortion, stereo manipulation, amp simulation and some very weird stuff. Their spartan user interfaces give them a retro feel, but they're an essential part of my sound design arsenal.

Derek Johnson

"There's some dross out there but there's also a lot of very clever programming happening outside commercial R&D"

UNIVERSAL AUDIO PULTEC PRO

This is the best Pultec emulation I've come across. Modelled on particularly famous hardware units, the Pultec Pro gives me both the EQP-1 and ME-5, all in one plug-in. I would certainly like a dollar (or even a cent for that matter) for every time I have called the Pultec Pro up on a session. It can really help infuse some lushness on otherwise bland and one-dimensional tracks. Either in your DAW as an insert or used in a stereo editor like Wavelab as a Tone/Deep/Air device, the Pultec Pro is a must. Sure, it doesn't have a lot of surgical control for notching out the nasties, but there are a zillion and one plugs that can do that. Tone, tone, tone, that's what the Pultec Pro is all about.

So good is the tone that you can quite easily overdo it. I often find myself backing off just a bit from where I like the tone when a track is soloed. My most common settings when using the Pultec Pro over the stereo bus are 1–3dB boost at 60 or 100Hz with a concurrent dip of 1–3dB, 1–3dB boost with a wide Q at 10, 12, or 16kHz, 1–3dB boost at 700Hz or 1kHz, a dip at varying areas depending on the source(usually 200Hz, 1kHz or 4kHz) and a boost at 1.5 or 2kHz.

I know it doesn't matter to the audiophile, but the GUI is just so damn good looking. In every way it replicates the simplicity and classic styling of the original hardware units.

Cal Orr

PITCH CORRECTION DISDAIN

I should explain first that in my small, backyard studio I usually work with musicians who are in, well, let's say the *lower end* of the talent spectrum. They have passion, drive and ambition by the bucketload to become the next Britney Spears (with hair) or James Blunt, and dress in clown suits and queue in the pouring rain for two days to get into *Australia Idol* auditions. Sadly, there's little actual *talent* involved and it can make things a bit tricky in front of a microphone.

So the plug-in I hate is anything that offers pitch correction, because when I once (in desperation) gave it a try, I got my fingers burnt. The vocalist I was recording immediately saw a Grammy on her mantelpiece. Pitch correction was the answer to all her woes – meaning her singing, which she'd just heard for the first time in crystal clean 48k, 24-bit clarity, instead of in the shower where she normally sang. It was disappointing, to say the least.

To head off a torrent of tears and having to hide everything sharp from within her reach, I hastily told her something like, "Don't panic! No one can really, *really* sing. They all use this nifty pitch-correction program. Look, I'll show you".

Yes, well, we all say some silly things at times and this definitely qualified as a moment of plug-in-driven, foot-in-mouth foolishness.

Thus I ended up spending countless hours trying to twist and bend a mediocre performance into some semblance of a tune. Each vocal take would finish not with the question – as you'd expect – "Was that okay?", but instead, "Will you be able to fix that?". To which I answered "yes" and promised to do later in my own time. Why? Because, well – look, before we go into *that* I agree pitch correction has its rightful place. But in extremes it irresponsibly provides hope for the hopeless, talent for the tuneless and – all right, the odd hit single for Cher.

We've already got virtual drummers, virtual guitarists and virtual bass players. Surely we have to draw the line at virtual talent? For goodness sake, before you know it, record companies and producers will be dragging totally unknown, tone-deaf teenagers off the streets and turning them into overnight sensations…

Maybe it's a bit late to do much about that.

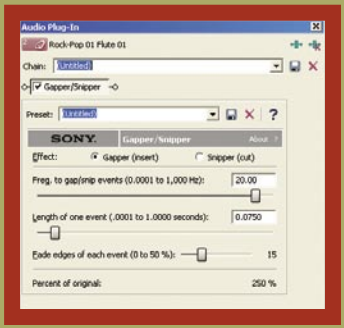
And okay, I'll admit it. The interface on pitch-correction plug-ins confuses the hell out of me. My one experience was made all the more torturous by having little idea what I was doing – but you can't tell the client that. That's why I did it later.

Bugger it, just give me a ping-pong delay anytime.

Graeme Hague

"'Don't panic! No one can really, *really* sing. They all use this nifty pitch-correction program. Look, I'll show you'."

Graeme Hague's running out of excuses. Software like Sonar 6 (pictured) comes with V-Vocal included.



"Technically, I have no idea what's going on. It's just a really cool anomaly."

SONY GAPPER/SNIPPER

In the early '90s I had an Ensoniq sampler, an ASR10. Anyone remember those? Every now and then you'd find a rogue sample file, one that wouldn't behave. You could load all the instruments with samples then load your rogue sample into the last instrument slot and, *bingo*, the rogue sample would randomly grab a tiny slice of each of the other loaded samples and play them all as one sample. I loved the haphazardness of those files.

I'm sure most folks were trashing these files, marking them down as 'corrupt data'. Not me, at the time they were treasure. My ASR10 is in one of my son's bedrooms now – a relic… last of the great workstations. These days, Sonic Foundry's (now Sony) *Gapper/Snipper* plug-in is my contemporary equivalent, a rogue sample player.

It works like this: you either set it to 'gap' your file or 'snip' your file. The Gapper inserts silence sections into your original wavefile at the frequency you specify. The addition of the slices of silence slows down the playback of your audio file. The Snipper slices portions out of your wavefile. Again, you choose the frequency and it speeds up your audio. I'll typically use it as a track insert. I generally find the effect doesn't immediately operate until several seconds after the wavesample is supposed to begin playing! And when I stop playback the effect still continues to bleed on. Technically, I have no idea what's going on. It's just a really cool anomaly. So here's one application: I like to record the effect and bounce it back into the original project. I'll re-time it so it sits nicely with the original file and, presto, I now have the ability to crossfade between the two tracks and create a sense of temporal dislocation, that the original file is de-constructing. Much fun!

Hugh Covill

PROTOOLS EQ

Just wanted to pay homage to the humble, basic old ProTools EQ plug-in. It doesn't do anything radical and is probably the least glamourous looking plug-in ever designed. That said, it's great for simple tidying up duties and I usually have at least a bunch of them on any mix – often rolling off a bit of bottom end or cutting an irritating frequency or doing a tad of high-shelf boosting. The main selling point being that it doesn't chew through your CPU headroom like a bastard and leaves you with enough juice to do something interesting elsewhere. Three cheers for those dull little men in the grey suits!

Greg Walker

TC NATIVE PARAMETRIC EQUALISER

The thing I find really useful about this plug-in is the 'Loudness' joystick control (see screenshot). This control is independent of the main parametric section and, once switched on, you can move the joystick around in any direction you like with the mouse pointer. This allows you to very quickly audition, at varying amounts, EQ curves that are similar (when boosting) to what the 'loudness' button on old hi-fi amps used to do: ie., boost both top and low end simultaneously. The joystick also allows you to cut both high and low frequencies as well as all the other permutations – in fact, 360 degrees of choice at varying amounts from the centre point. As best as I can tell from the smallish EQ display, the curve is centred around 1kHz, providing a gradual increase that peaks at about 8dB at 20kHz and 20Hz.

I often have to do a quick EQ on a reference mix, and mostly this plug-in does a great job of providing an overall curve that can give a dull track some life as well as indicating what the track might need EQ-wise in mastering. The actual parametric section is also very powerful, offering seven control sections with selectable parametric, low/high shelf and notch filters. The stereo sections can be unlinked to give control over both left and right signals, which are reflected as separate, coloured graphs in the EQ display. You can also combine the parametric and loudness controls to give you 'off the screen' amounts of EQ, and on top of that there's the 'Soft Sat' valve/tape emulator. Unfortunately, this only has an on/off switch – which is a shame really, as having control over other aspects of this process (like threshold and gain) would have been great. As it stands, on tracks that have not been compressed to the max, this saturator can give warmth and density to a mix. There is reference to the TC Native Bundle in the discontinued section on TC's website, so if you were looking for a version you'd have to get one secondhand.

Robin Gist

ALGORITHMIX RENOVATOR

Like many DAW owners, I probably use more plug-ins than I really need. The reality is that the processing algorithms that come 'for free' as part of today's DAW host applications are usually very good, but are often sorely neglected by users lured by attractive plug-in GUIs and the often fanciful notion that other plug-ins are somehow superior. That said, plug-in architecture offers developers the opportunity to code specialist algorithms, and some have proven to be very good at staking out unique turf. My favourite plug-in developer is a German company named Algorithmix, makers of class-leading EQ and restoration plug-ins. Choosing my favourite plug-in was a toss up between various Algorithmix tools (the Blue analogue modelLing EQ is a current fave), but I simply couldn't do a lot of my work nowadays without Algorithmix ReNOVAtor running within Sequoia.

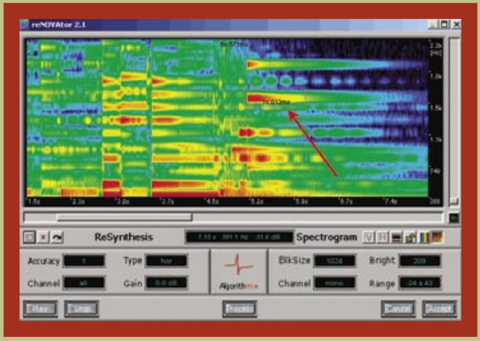
Comparable in some respects to Cedar's ReTouch, ReNOVAtor allows transparent removal of audio anomalies without affecting the remaining audio. These two expensive but awesome plug-ins were unique when they appeared a few years ago, although they have recently 'inspired' a few competitors to develop similar tools (as one example, Magix has now developed a similar but less powerful tool which runs natively within Samplitude and Sequoia). The power of ReNOVAtor astonished me the first time I used it and still astounds me on a daily basis.

Using a brilliantly designed, very flexible graphical interface, ReNOVAtor displays a frequency-based spectrum across the time domain, allowing the user to easily identify audio anomalies by colour. The user then selects the anomalous region and clicks a Process button. ReNOVAtor removes the anomaly and replaces the removed material with interpolated data from the surrounding audio. With over a decade as a leading developer of high-end restoration software and linear phase processing algorithms, Algorithmix has definitively shown that remarkable things can be achieved with high quality interpolation, especially at ReNOVAtor's double precision (up to 80-bit) floating point resolution. The beautiful (even miraculous) aspect of ReNOVAtor's superb interpolation is the capacity to replace damaged audio without resorting to the traditional time consuming edits and kludges, whereby experienced engineers used an array of techniques to obtain results which were rarely transparent. Those days are gone.

Need to remove a cough from a quiet violin concerto, without damaging the violins *at all*? ReNOVAtor does this without even trying. Vocal 'plosives, guitar string squeaks, clicks, pops, hiss, chair noises, thumps, bumps – these are all a doddle for this plug-in to remove without a trace. Cleaning up fades has never been easier. But ReNOVAtor takes this paradigm a step further, offering identification and removal of offensive harmonic distortion. Simply identify the fundamental – ReNOVAtor finds the rest… Hearing/seeing this plug-in deal with such things as earth hum is a jaw dropping experience! Anyone working in mastering, restoration, broadcast, or post/film will love this plug-in, but it's also very useful within typical multitrack mix environments. Not cheap, but worth every cent.

ReNOVAtor plugs into ProTools, Wavelab, Soundscape, Sequoia and Pyramix. It is now available for Mac users and is also available as a standalone processor.

Sean Diggins



"Hearing/seeing this plug-in deal with such things as earth hum is a jaw dropping experience!"

APOGEE UV-22

It comes in many different shapes, it's based on a psychoacoustic concept few people truly understand and, by definition, adds more noise to your record than any other process you'll ever use (often without you even knowing it or hearing it). The plug-in I'm talking about is Apogee's UV-22 dither plug-in.

That's not exactly a glowing introduction, but this plug-in is hailed by top-level mastering engineers around the world for its ability to add depth and clarity to a mix. Its benefits range from 'rounding out the rough edges of digital', to boasting 'near 20-bit resolution being heard from a 16-bit CD', to making 'already-mastered 16-bit sources sound better'.

I use this process more than any other in the mastering studio; in fact, every project set for CD release I've ever worked on has been passed through the UV-22 plug-in. It's been estimated that as many as eight out of 10 hit records in the US are mastered with UV-22.

I currently use the Apogee's UV-22 plug-in as an insert process in SADiE when mastering. UV22 is also included in many of Apogee's hardware converters.

Mark Bassett