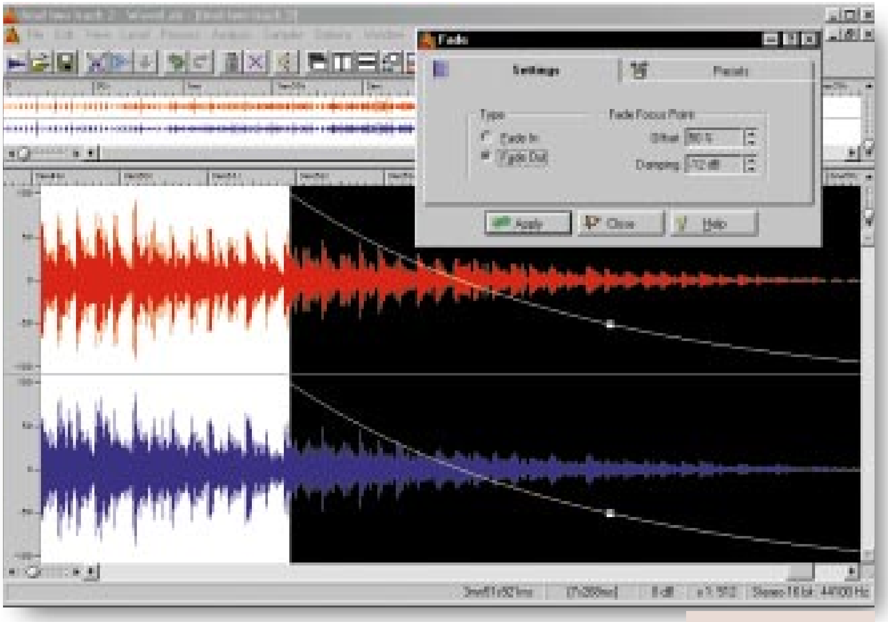
Do the write thing

Writing to CD is now cheap and easy. Steven Helstrip goes on record.

Your home-grown recordings should be sounding heaps better if all the suggestions from the past three *Sound* columns have been put to good use — it just takes a hint of compression and some EQ to make a world of difference. This month, as promised, I am going to look at how to get those recordings onto CD as cleanly as possible. I have also taken Yamaha's newest rewritable drive for a spin.



▲ **Fig 1** To create a gentle logarithmic fade in Wavelab, try an offset of 50 percent with -12dB damping

Until relatively recently, producing a CD required some serious cash investment. These days, however, you can get your hands on a CD writer for less than 200 sheets — about the price of a decent cassette recorder.

Strange though it may be, blank CD-Rs are now cheaper than most audio cassettes and offer far greater flexibility. When I master a CD, for instance, I create a mixed mode disc and archive my sequencer arrangements, audio files, softsynth patches and even an image of the CD inlay to the data partition. This leaves plenty of room on my hard disk to start a new project. So how is it done?

First steps

Once you have a mix you're happy with as audio tracks in your sequencer (*refer back to the February issue column for the full low-down*) listen to each track in turn and remove any unwanted noise such as background hum from microphone recordings.

You can get your hands on a CD writer for less than 200 sheets

If you have a noise gate plug-in, multiple instances can be inserted on the worst offending channels until your PC runs out of steam. Alternatively, snip out the noisy sections by using the scissors tool, paying particular attention to quiet passages.

If you're planning to fade out your track towards the end, don't worry about inserting volume envelopes in your sequencer; this is far easier to achieve with an audio editor once the arrangement has been recorded as a standard stereo wave file. Most audio programs have a 'mixdown to stereo' facility that does this for you: in Cubase, set the locators to the start and end of your song and select Export Audio from the file menu; in Cakewalk, the command is Mixdown Audio, which can be found under Tools.

To be compatible for CD audio, the wave file needs to be saved in 16-bit stereo with a 44.1kHz sampling rate. If you have used sequencer effects and automation, these options should also be checked in the mixdown windows.

Once the mixdown is complete, listen back to ensure that no unusual side effects have crept in and load the track into your sound editor. From there you can apply a fade-out if necessary and trim the intro. Remember, though, that some CD players don't instantly spring into play mode so it's best to leave around 50ms of silence just before your audio begins, otherwise, you may lose a slice from the beginning of your songs when skipping from track to track. When it comes to applying the fade, bear in mind that a logarithmic curve will sound more natural than a straight linear fade [Fig 1].

I generally set the fade over six or seven seconds depending on the material. To ensure as much signal goes to CD as possible, normalise the track to 0dB before saving. You may even consider applying compression to the overall track or a master EQ. Given the effects of compressions, though, it's best to do this before applying a fade.

Writes and wrongs

Unlike data CDs, audio discs need to be written in one go (or session). But if anything should go wrong before the session is closed the disc will be rendered useless. So that you don't waste more discs than necessary, I have outlined below the potential pitfalls and how best to avoid them.

• One of the main reasons why disc writes fail is due to contamination of the disc surface; it only takes a speck of dust and it's 'game over, man'. To avoid this, don't remove a blank CD-R from its case until needed and, just as importantly, never touch the write surface.

- CD writers have a buffer (usually 1 or 2Mb) where data is stored before being written to disc. If your PC doesn't keep up with the recorder, the buffer runs empty; the dreaded buffer under-run. If this error message is reported, and it's not that uncommon, try disabling any programs that may interrupt the writing phase. These include screensavers, power management utilities, CD auto insert notification and even fax software.
- Burning CDs requires a continuous stream of uninterrupted data, so you should ensure that your hard drives are defragmented. Do not use your PC for anything else while writing is in progress. But if all else fails, try reducing the write speed from 4X to 2X, or from 2X to 1X.

YAMAHA CRW4416SX DRIVE

RECOMMENDED Personal Computer World

Yamaha introduced the first ever recordable CD drive back in 1989, so it comes as no surprise that it leads the way today. The 4416 is a tray-loading device capable of writing and rewriting at quad speed, while playback is a respectable 16-speed. It comes in either SCSI or IDE configurations and there's an external model for SCSI, which is the drive under scrutiny here. Although the most expensive, external drives have a big advantage: they can be switched off when not in use, which prevents them from overheating.

The external case is only just bigger than the drive itself and provides a switch for power, SCSI ID select, and two SCSI 2 connectors. The first thing I noticed about the 4416 is how quiet it is; other drives I have used sound like they might take off once they reach 16 speed. As you would expect, all the main CD formats and writing modes are supported and there's an essential 2Mb buffer to prevent under-runs. In the six weeks I've been using this drive, I have had a 100 percent success rate.

The package includes one CD-R, one rewritable disc and a copy of Adaptec's Easy CD Creator. With a typical street price of £349 it's not the cheapest drive around, but if you're looking for a solid workhorse I'd highly recommend it.

Price *£410 (£349 ex VAT)*
Contact *Yamaha 01908 368872*

Questions & answers

Q I have been using Cubase VST for MIDI and audio sequencing for several months without problems but since having upgraded my AWE-64 Gold to a SoundBlaster Live! I'm having trouble with synchronisation. When I playback my Cubase arrangements, audio and MIDI start in sync but drift apart after about a minute into the track. I have experimented with different buffer sizes and have increased the pre-roll setting in the Sync menu but to no avail. Have I missed something?

NEIL PERRING

A ***Cubase uses your sound card's digital clock as a time reference to sync MIDI tracks to audio. From what you've described, it sounds as if the problem is related to the SB Live!'s clock output, which is fixed to 48kHz. If your previous arrangements were recorded at the default 44.1kHz sampling rate, drifting will occur over time. The first thing you need to do is set Cubase to operate at 48kHz to match the sound card (Audio System Setup dialogue — see Fig 2). This will ensure that new songs sync-up correctly. To play back your old arrangements in sync, you will need to convert, or resample, the audio tracks to 48kHz and reload them into your songs. Most audio editors have this facility.***

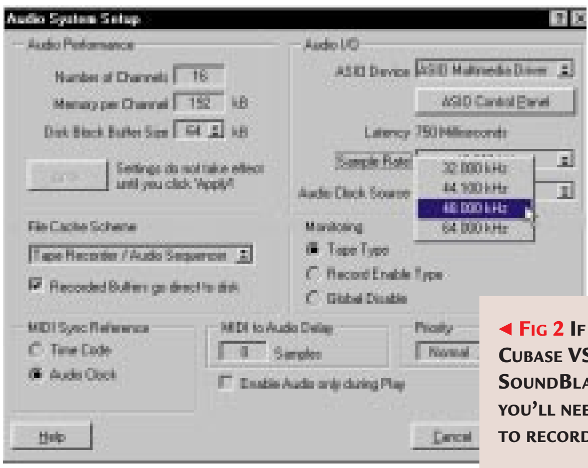


Fig 2 If you use Cubase VST with a SoundBlaster Live, you'll need to set VST to record at 48KHz

PCW CONTACTS

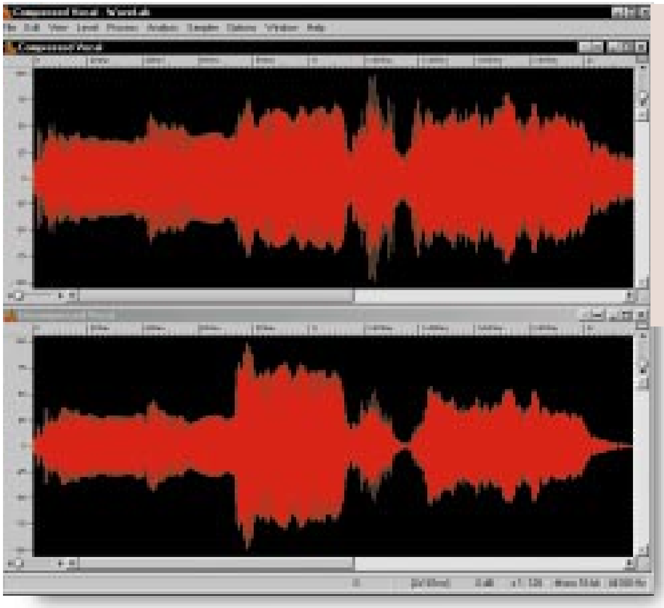
Steven Helstrip welcomes your feedback on the Sound column; it's music to his ears. Contact him via the PCW editorial office (address, p14) or email sound@pcw.co.uk

Create a good compression

Steven Helstrip explains the creative force of the useful compressor tool for smoother tracks.

Over the past two months we've discussed ways of using EQ to improve the clarity of your mixes. Hopefully by now you've had a chance to try out some of the examples (not to mention those free plug-ins) and are ready to tackle what is probably the second most useful audio processing tool, the compressor.

So what do compressors do? In simple terms, when audio is compressed all the loud sections are attenuated, or reduced in level, while the quieter sections are raised in volume. So really, a compressor is like an automatic volume control that keeps an eye on the input gain of a signal and makes the necessary adjustments to output sound on a more even, or constant, level. Fig1 illustrates the effects of compression on a vocal recording. The settings used are quite extreme, but illustrate what goes on.



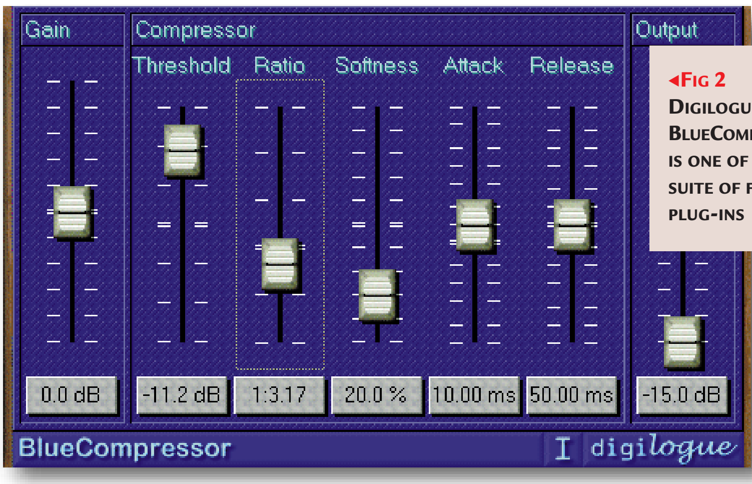
◄Fig 1 The waveform displayed in the lower half of the screen shows the natural dynamic range of a vocal recording. In the upper half of the screen, the vocal has been compressed

Compressors are insert-type signal processors. As such, they can be used on individual instruments in a mix, or across the main mix output to smooth out an entire track. Once audio has been compressed it has less dynamic range, which enables you to raise its overall level. In fact, gain make-up controls can be found on the majority of compressor units and plug-ins.

Compressors are not only used to control the overall dynamics of a track, though. They have many creative uses in the studio, virtual or otherwise, and enable you, say, to extend the natural decay of instruments and add punch, or greater impact, to percussive sounds. In a similar way to EQ, if you have a basic grasp of using compression you can greatly improve how your music sounds.

A compressor is like an automatic volume control

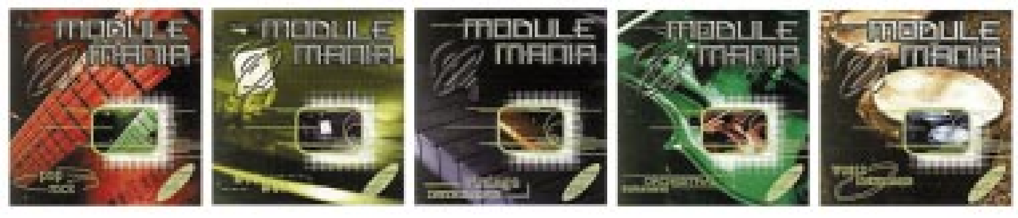
Now that we know what they are, let's take a look at how they work. Compressors come into effect when the input signal rises above a set threshold. The amount of compression applied — also called gain reduction — is determined by the ratio setting. With a ratio of 1:1, the sound will remain unchanged because for every 1dB of input level that rises above the threshold, 1dB of level will pass through to the output stage. A setting of 6:1, however, means that for every 6dB of sound which rises above the threshold, the output will only increase by 1dB so the higher the ratio, the stronger the effect.



◄Fig 2 Digilogue's BlueCompressor is one of 11 in a suite of free plug-ins

Attack and release are two more controls which greatly affect how compressors work. Attack determines how quickly the compressor comes into effect once the threshold has been passed. With a setting of, say, 1-3ms the compressor will kick into action almost immediately. When audio is attenuated this quickly it comes across with more punch and sounds crisper. With a setting of around 400ms the effects of compression are more subtle. The release control sets how long it takes for the output to return to its normal level once the input signal has fallen below the threshold.

Depending on the instruments and the type of music with which you are working, compressors can have many different effects. Moreover, each control interacts differently with the others depending on how they are set. Confusing, or what?



E-MU MODULE MANIA SOUNDFONT CDS

E-mu has released five of its professional synthesiser modules in SoundFont format. The collection includes the Planet Phat, Vintage Keys and all three Proteus expanders. Each synth has featured on countless top ten hits over the years and can now be yours for just £25 apiece, or £80 for the lot. Here's a quick run down of what's what.

Proteus I is best suited to pop music and includes a classic selection of pianos, strings, organs, brass and percussion. To give you some idea of their quality, there is 30Mb of instrument samples comprising just 128 presets. In contrast, most PC sound cards have just 2Mb of instruments stored in ROM. **Proteus II** has over 40Mb of orchestral sounds, covering everything from solo flutes and oboes, to marcato and tremolo string sections.

Proteus III is dedicated to world instruments and contains a slightly more obscure set of sounds. Examples of what's in store include gamelan, Irish harps and Asian gongs.

Planet Phat was designed for hip-hop and R&B, providing a wealth of cutting-edge bass, drum and guitar samples.

Vintage Keys is perfect for dance and includes over 220 analogue synths.

Contact E-Mu on 0131 6536556 or at www.emu.com. For a wide selection of free SoundFonts go to www.sblive.com.

Hot utilities on our cover CD

Two new utilities have become indispensable to my work so I feel duty bound to tell you about them.

The first is a replacement CD-ROM device driver for Windows 95/98 which enables audio tracks (CD-DA) to be viewed as wave files from the Explorer. For example it enables you to load CD audio tracks straight into Sound Forge or WaveLab for editing, replacing the need for a dedicated audio ripper. The driver should work with any CD drive supported by Windows — I can't vouch for this although I have tested it with three CD drives with great success. The driver (CDFS.VXD) can be found in the handson\software\sound folder on this month's cover CD. To install it, copy it into your Windows\System\IOSubSys directory and reboot. You may want to backup your original CDFS.VXD first, to be on the safe side.

The second utility is Virtual Audio Cable, or VAC for short. This is a driver that enables audio to be routed from one application to another in much the same way that Hubi's LoopBack Device allows you to interconnect MIDI programs. So, for example, you can record the output of a software synth straight into your sequencer or record the output of an effects processor to your sound editor. The driver is multi-client so any number of applications can simultaneously access your sound card. VAC can also be found on this month's cover-mounted CD. Have fun.

▲**Fig 3** Synchromesh's KwikKomp 3 is a doddle to use and packs quite a punch

Setting up

There are no right or wrong ways to dial-in your own settings. Neither are there any rules when it comes to deciding what gets compressed and by how much. If the track in question sounds good and cuts through with compression, then you must have set it up right — that's all there is to it.

As a rough guideline, I tend to get started by setting the ratio to around 4:1. Next, I loop a fairly quiet section of the track and lower the threshold until there is just a touch of gain reduction — most compressors have a meter to show reduction levels. Then I adjust the attack time, listening carefully to the effect on the audio. When the compressor is doing its job quickly enough without being too obtrusive, I tweak the release until there's a natural, even-sounding decay.

It is important to A/B the results with the original mix by occasionally by-passing the effect. And by all means, if you find that compression does not help to bring out the best in your music, then don't use it.

Tip: Set your compressor before reaching for EQ as compression changes the tonal characteristics of a sound. With any luck, you may not need to use EQ at all.

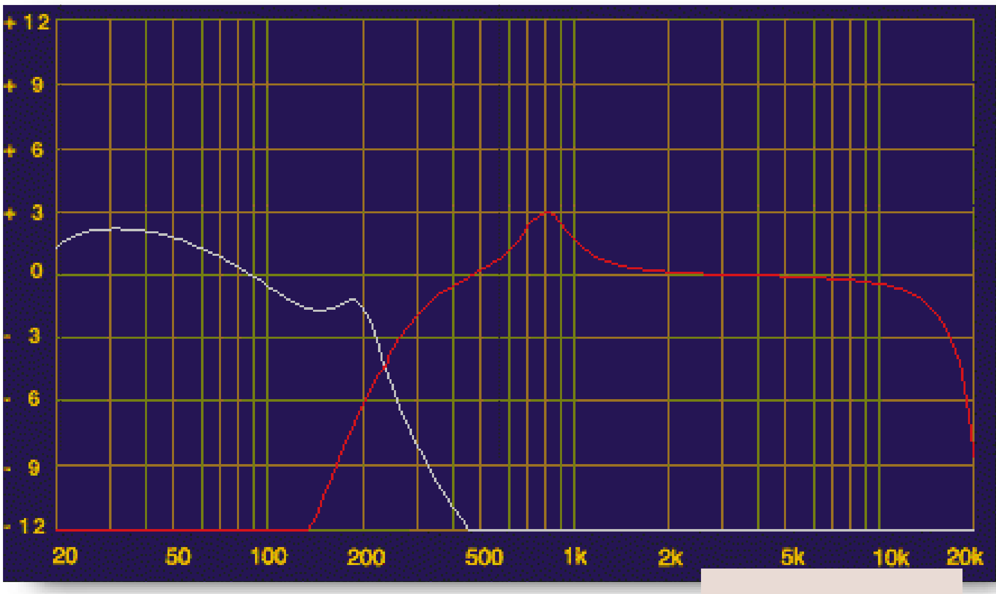
If you don't have a compressor installed on your PC, there are two free plug-ins available on the internet for VST and DirectX-compatible applications: BlueCompressor [Fig 2] and KwikKomp 3 [Fig 3]. These can be downloaded via the Cubase web ring. Go to www.webring.org/cgi-bin/webring and just type Cubase.

PCW CONTACTS

Steven Helstrip can be contacted via the PCW editorial office (address, p14) or email sound@pcw.co.uk

Sitting comfortably

Use EQ to make a sound sit well in your mix. Steven Helstrip explains.



▲Fig 1 EQing two bass-type sounds

To continue our mini-series on mixing audio, we're going to look at EQ in more detail and explain where, when and how to use it. Although EQ can be used as a creative tool, its main purpose is to shape and control a sound so that it 'sits' well with other instruments in a mix. But what does that mean, exactly? When mixing a track, what we're aiming to do is to make each instrument come across as clearly as possible over the speakers. We can do this with EQ by separating (or partitioning) instruments into their own spaces within the audio spectrum. Of course, there's a good deal more to mixing than EQ alone, but this is a good place to start.

The more instruments or parts you have playing in a song, the more difficult it becomes to mix them. We can illustrate the problem with this simplified example. If two untreated bass instruments play simultaneously, they are likely to sound cluttered as, by their very nature, they occupy similar frequencies in the audio spectrum. However, they can work together well if they are separated into their own spaces.

There are three ways to achieve this effectively. The most obvious is not to play them in the same place. It's often overlooked as a basic principle of production — less can be more. The second approach is to pan them to opposite speakers, although this may not always be appropriate. If neither of these work, then EQ has to come into play.

Fig 1 shows how two bass-type sounds might be EQed to fit into separate frequency ranges. The red EQ curve has had most of its bass frequencies filtered out (or rolled off) to allow the instrument, represented by the blue curve, space to breath. There is some area of overlap but this is perfectly alright given that it's a relatively small range. Conversely, the blue curve has had much of its *higher* frequencies filtered out. In addition, you can see that each curve has a bell-shaped peak where gain has been applied to bring out a particular quality in each instrument's tone.

The main purpose of EQ is to shape and control a sound

The art of mixing can take years to master but you can improve the sound of your music with a basic understanding of EQ. By far the best way to learn is to get some direct hands-on experience. So what are you waiting for? Here are ten general guidelines to help you get started. And if you need an EQ or filter plug-in, we've got that covered, too, in the panel overleaf.

1 Before you reach for the EQ, ensure that you have done everything possible in the recording process to get the right sound onto disc in the first place. Careful sound selection can keep the use of EQ to a minimum.

2 Don't spend too long EQing an instrument in isolation as it will sound totally different once it's back in the mix.

3 Cutting frequencies you don't want, rather than boosting those that you do, can be just as effective and helps to open up a sound.

4 Listen carefully to how commercial tracks are mixed. This is by far the best way to learn how to approach your own mixes. Also, take time out to practice and experiment.

5 To add a click to a kick drum, boost the frequencies around 6KHz. To help it 'bite' through the mix, boost around 2KHz.

6 To add clarity to a bass instrument, cut the frequencies around 250-300Hz. A boost of around 100Hz will add body and weight.

7 Take the abrasion off hi-hats by cutting from the 1-2KHz range. Meanwhile, sparkle can be added by boosting the frequencies around 10-12KHz.

8 Vocals generally sound warmer when you cut around 1KHz, while boosting the 2-3KHz range adds presence.

9 To make pads sit back in a mix, cut the frequencies around 2-3KHz. A boost around 8KHz will add clarity.

10 Leave any final EQing until the following morning if you can. Your ears will be fresh and any problems will be easier to spot.

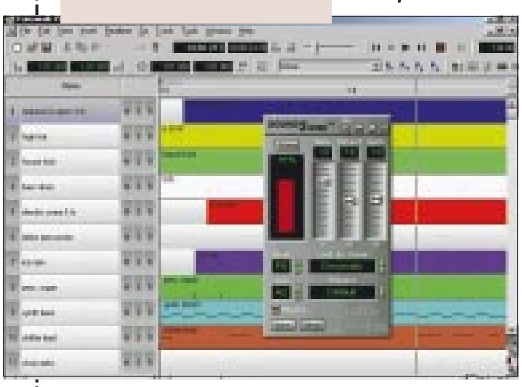
Questions & answers

Q I have a problem recording melody lines into my sequencer because my keyboard skills are minimal. Entering notes in step time is not satisfactory as there is no natural flow. Do you know of a program which would take an input from a microphone and convert it to MIDI so that I could sing, hum or whistle the melody and then assign an instrument to it? I can string a few chords together and have no problem with bass lines but a naturally flowing melody defeats me.

BERNARD MANTELL

a *There is a program called Sound 2 MIDI which will do just that. It runs alongside your sequencer and can take an input from either mic or line inputs. Once it knows the key and the mode of the music it's about to transcribe [Fig 4] all you have to do is hit record in your sequencer and, er, sing. I just hope you can sing in tune! See the PCW Contacts box for details.*

▼FIG 4 SOUND 2 MIDI TAKES AN ANALOGUE SIGNAL FROM YOUR SOUND CARD'S EXTERNAL INPUT AND CONVERTS IT TO MIDI



Q I have some old four-track demo tapes that I'd like to transfer to my PC for digital editing. Do I need a dedicated four-input sound card to record them or can I simply buy a cheaper, second stereo input card to use with my Sound Blaster 16?

PAUL WARD

a *The problem with using two separate cards for recording and playing back multiple channels is that each card's audio clock source may run at different speeds. Although we're only talking about tiny fractions, over a few minutes the two cards will almost certainly drift out of sync, even if they're the same type. From what you have described, though, you don't actually need four inputs. Simply record the tracks off tape two at a time and use your audio editor to line up the separate takes — that's the beauty of digital editing.*

Q Do you know of a program that can convert and compress to and from various audio formats and has a batch mode (like Paint Shop Pro)?

ALEX HELFET

a *Two shareware programs spring to mind. The first is Convert, a DOS-based command-line utility. It supports around 50 audio file formats and provides basic batch processing. It's straightforward to use but if you'd prefer a Windows application, check out Awave. which supports nearly 200 file types including SoundFonts and Mpeg. But it doesn't have the batch processing facility. So, best get both, then. These programs are available from www.maz-sound.com.*

DIGILOGUE BLUELINE PLUG-INS

There's no shortage of free plug-ins on the internet but I've never before come across anything so complete as this. BlueLine is a suite of 11 plug-ins for VST and DirectX-compatible applications. It's a great set but the two in which we're particularly interested are filter and EQ modules [Figs 2 & 3]. Together, they provide everything you need to start mixing, including four fully-parametric bands on EQ and no less than 16 filter types. Although both are big on features, they actually use very little processor overhead and should work with any system, so now you've got no excuse!

▲FIG 3 THE BLUELINE FILTER OFFERS 16 FILTER TYPES WITH LEVEL TRACKING

▲FIG 2 DIGILOGUE'S EQ MODULE PROVIDES FOUR FULLY-PARAMETRIC BANDS WITH ADJUSTABLE Q

PCW CONTACTS

Steven Helstrip can be contacted via the PCW editorial office (address, p14) or email sound@pcw.co.uk

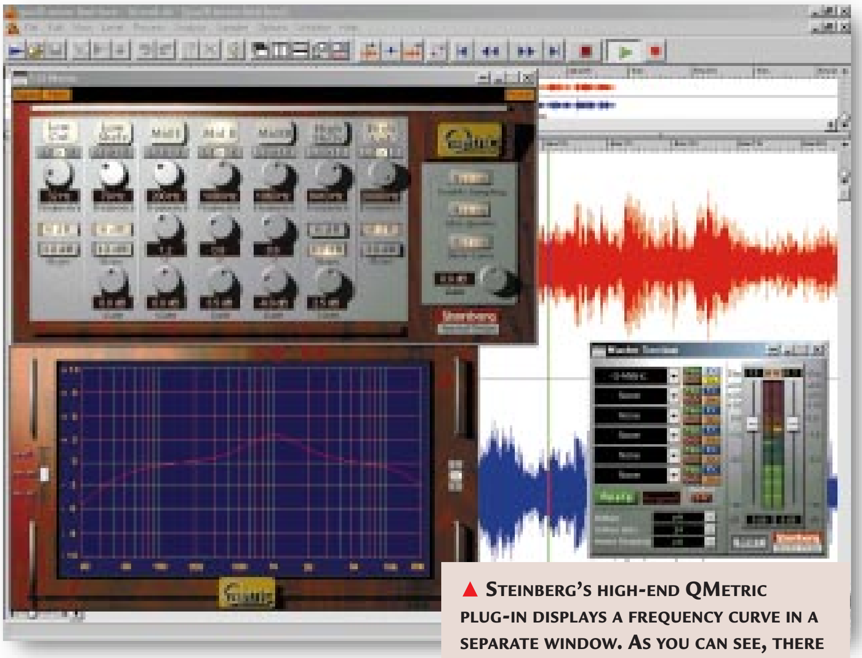
Digilogue BlueLine plug-ins http://members.tripod.de/digilogue/

Sound 2 MIDI £99 (£84.26 ex VAT) from Et Cetera Distribution 01706 228 039

On an equal footing

Improve your mixes no end with a touch of EQ. Steven Helstrip shows you how.

We certainly struck a chord in last month's column: feedback has been very favourable. To recap, we started to look at the processes involved in recording, mixing and mastering audio to CD. So far, we've covered how to record your MIDI instrument tracks back into your sequencer as wave files, which will enable you to make a two-track audio master. But before we reach that final stage, there's a vast range of effects that can be used to add a touch of professionalism to those home-made recordings. So, to start with we're going to look at EQ, the most commonly used effect of them all.

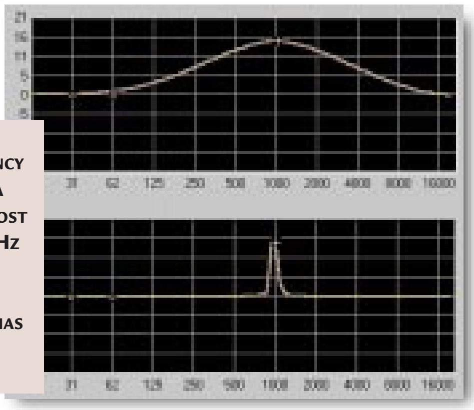


▲ STEINBERG'S HIGH-END QMETRIC PLUG-IN DISPLAYS A FREQUENCY CURVE IN A SEPARATE WINDOW. AS YOU CAN SEE, THERE IS A FREQUENCY BOOST AROUND 1KHZ, WHILE THE LOW AND HIGH FREQUENCIES HAVE BEEN ROLLED OFF SLIGHTLY USING LOW- AND HIGH-CUT FILTERS

EQ, or equalisation, was originally designed as a corrective effect to make up for the loss of frequencies in early recordings. What started out as basic tone controls (similar to bass and treble on a home stereo system) have developed to become precise and creative studio tools. In reality, an EQ module can do only two things: increase (boost) or reduce (cut) audio frequencies. Yet it can take a lifetime to master those few controls, especially when it comes to mixing 12 or more separate audio channels. But with just a basic grasp of EQ-ing, you can considerably improve your mixes.

EQ modules fall into two categories, graphic and parametric. A graphic EQ may have up to 30 preset frequency bands which you can modify with faders [Fig 1]. These bands will typically start from around 40Hz (sub-bass regions) and rise up through the full frequency spectrum to around 16kHz. Parametric EQs enable you to dial-in specific frequencies and they provide at least two controls, frequency and gain. The frequency control is used to select the frequency range you want to modify. The gain setting then enables you to cut or boost that range. Some EQs provide a third control, called Q, used to set the bandwidth of the frequency range [Fig 2]. The higher the number, the narrower the range.

▶ FIG 1 TC WORKS' NATIVE PARAMETRIC EQ ENABLES YOU TO SELECT BETWEEN 7, 14 AND 28 BANDS OF EQ. TO APPLY CUT OR BOOST YOU SIMPLY DRAW A CURVE ONTO THE TOUCH-SCREEN



▶ FIG 2 THE UPPER FREQUENCY CURVE SHOWS A FREQUENCY BOOST AROUND 1000HZ WITH A LOW Q SETTING. THE LOWER CURVE HAS A HIGHER Q SETTING

EQ is used to create separation in a crowded mix by using it to narrow the frequency range an instrument occupies. It may also be used to create special effects. For instance, you can make a vocal sound as though it's being sung over a telephone by limiting its frequency range (or band) to that of a telephone (around 1kHz), or you may simply need to boost the bass or treble frequencies to add weight or presence to a mix.

There are no hard rules on how to apply EQ. If a setting sounds good, then it is good. Let your ears be the judge.

Questions & answers

Q Just finished your bit in December's issue — lots of tips, cheers. I have recently upgraded my PC to a 266MHz Pentium II, which includes a Matrox Productiva graphics card. Everything works a treat, but the timing in Cubase seems to be messed up slightly. Would you know what the culprit is likely to be? Also, I'm thinking of upgrading to a 380MHz AMD K6-2 with a 100MHz bus and lots of 100MHz RAM. The big question is: does the AMD K6-2 work better or worse than a PII? One last thing (honest!) — the 512K RAM on my AWE 32 is so pitiful that I can only load drum beats and high hats into it. What sound card would you recommend I upgrade to?

WILL SHAND

***a** There are many things that can cause timing problems with Cubase, but once you've got to the bottom of it, you should have a solid system sitting in front of you. Hiding away on your Cubase CD-ROM is a troubleshooting Acrobat file (or PDF) which has a large section on MIDI timing. If it's not covered in there, check out Steinberg's Knowledge Base on the internet. With regard to which processors are best, Steinberg still recommends Intel chips due to their higher floating-point performance — crucial for audio processing. As to which sound card should you buy, well, at just £130, nothing around at the moment comes close to Creative's SoundBlaster Live. Not only does it sound great, but you can also use up to 32Mb of your PC's system RAM for sampling. You can find the Steinberg Knowledge Base at www.ca.steinberg.net/main.html.*

EQ tips and tricks

Here are some general equalisation guidelines which may help you to achieve your musical goals more quickly:

Before reaching for the EQ dials, first try to hear in your head how you want the instrument to sound. Listen to commercial mixes and compare them to your own to see if you're on the right track (no pun intended).

If you're trying to remove an element from a sound, such as hiss, visualise where in the frequency spectrum it lies.

Next, increase the gain, say 8dB, and then sweep through the frequency spectrum until the sound you want to remove becomes as pronounced and obvious as possible.

Following this, reduce the gain. The exact amount of cut should be decided by listening to the part in relation to your mix.

When attempting to 'bring out', or enhance, a particular element within a sound, the natural inclination is to tune in to the sweet spot and then boost the gain. However, you can reach a point where you boost many bands; in real terms, you may as well just turn the whole track up. It's better to cut the frequencies you don't want. This will open up the sound and create 'space' for other instruments.

◀VST's BUILT-IN EQ SECTION. HERE, IT IS SET UP TO APPLY 2.9DB GAIN AT APPROXIMATELY 6KHZ TO MAKE THE KICK-DRUM TRACK MORE CLICKY

WAVEPLANT SAMPLE CD

Reader Ben Rossborough has spent much of the past four years developing a one-off analogue synthesiser which he calls the WavePlant. It's unique in every sense and comprises, among other technical wizardry, nine sound oscillators, six LFOs, nine envelope generators and six banks of filters. Roughly translated, that's a lot of sound-creation potential, 140 examples of which have been captured on this sample CD. Listening to sample CDs is no more exciting than looking through a book of typefaces, but when something refreshing comes along, it makes the whole experience that much more enjoyable. There are three examples featured on this month's *PCW* cover-mounted disc, so do have a listen.

The WavePlant can certainly be described as different, although there are some sounds which are reminiscent of the classic VCS 3. The disc starts with a collection of bass samples including sounds that range from deep and clicky, dubby and squelchy, though to Moog-like and... well, the unusual. Then we go on to effects, organs, synth leads and a selection of altogether off-the-wall synth textures. You get most sounds in audio, wave and Sound Font format, both with and without effects (usually reverb or distortion).

If you're looking for an all-round collection of dance sounds and breaks, this isn't for you. If, however, you want to get your hands on a collection of synthesised sounds that have never been heard before, this is a great disc to add to your collection, although you don't get too many sounds for your pounds.

Price *£25 (inc VAT and delivery)*
Contact Ben Rossborough
01497 820134

PCW CONTACTS

Steven Helstrip *can be contacted via the PCW editorial office (address, p10) or email sound@pcw.co.uk*

Playing at home

Record, edit and master your own CDs at home. Steven Helstrip shows you how.

Over the next few months we're going to tackle the basics of recording, editing and mastering your own audio CDs at home. On the way, we'll look at how to get the best from your sound card and how to record synths into your sequencer, and we'll include hints and tips to help you polish your home recordings. Even if you don't have access to a CD writer, there will still be lots here to interest you.

Transferring MIDI tracks to audio tracks

To prepare a two-track master which can be used to write an audio CD, MIDI tracks from your sequencer, including those played from a sound card, need to be recorded to your PC as audio files. If your music is purely MIDI-based (i.e. there are no sequenced audio tracks) this is pretty straightforward.

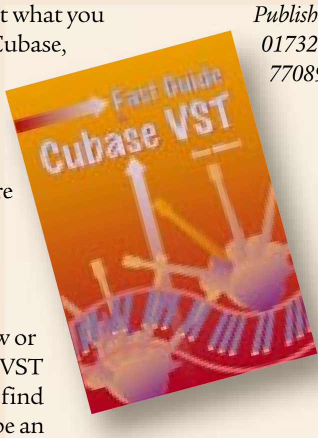
- **Using your sound card's** mixer utility, select Record Options and mute all the inputs except for MIDI.
- **Simply set** your wave editor into Record mode and play back your MIDI sequence. This will record all FM, WaveTable and SoundFont instruments.
- **Because we're preparing** an audio file for a CD, ensure the sampling rate is set to 44.1kHz 16-bit stereo. More than likely, you'll have a couple of audio tracks lined up in your sequencer and will need to transfer your MIDI tracks over. Not only will this allow you to create a two-track mixdown of your song, but you'll also be able to process MIDI parts with plug-in effects and EQ.

For maximum flexibility, record MIDI instruments to as many separate tracks as possible. You can always bounce, say, all the drums down to a stereo track once you have a mix you're happy with.

MIDI instruments can be recorded to audio tracks using the same approach we've just covered. However, if you also have external synths playing, these must be connected to your sound card's line

FAST GUIDE TO CUBASE VST

Cubase has come such a long way over the years that sometimes I wonder how newcomers to the program ever get off the ground. MIDI and audio are complex enough issues by themselves, but when they're thrown together in a full-blown professional package, where do you turn for help? Whether you're looking for quickstart tutorials or involved in mixing projects, the *Fast Guide to Cubase VST* could be just what you're looking for. The book covers most MIDI and audio topics in detail, and includes excellent chapters on Logical Edit, audio editing and mixing with VST. The author, Simon Millward, is clearly a proficient Cubase user and has plenty of sound advice (no pun intended) to offer, including the use of plug-in effects and EQ, through to choosing and setting up the right sound card. It's not just about what you can do with Cubase, though. The program's limitations are clearly revealed, as are the potential problems you're likely to encounter. If you're a new or intermediate VST user, you will find this book to be an excellent supplement to the online Cubase manuals. It is well written and packed with practical tips, although there really wasn't any need to include another list of general MIDI instruments.



Price *£19.95 (inc VAT)*
Contact *PC Publishing 01732 770893*

NEW — SOUNDBLASTER LIVE

I recently checked out the new SoundBlaster Live sound card. I was so impressed with its output quality that I installed an Event Gina, a £500 professional audio card, alongside it to do an A/B comparison. For the various listening tests I monitored each card's performance over a pair of Tannoy studio reference monitors. Suffice it to say, there was little to set the two cards apart. The Live has a very clean and detailed output with virtually no discernible levels of noise. Add to this 256 voices of polyphony from its three onboard synthesisers, and the ability to use up to 32Mb of your PC's system RAM for sampling, and you have a very desirable card for making music with your PC. Given the asking price of just £130, I'd recommend it highly.

Contact *Creative Labs 0118 934 4744*
www.soundblaster.com

Winner of Terratec Home Studio

In the September '98 column we ran a competition to give away a Terratec Home Studio, comprising an EWS 64S sound card and a full-size, four-octave MIDI controller keyboard. We were swamped with entries, but the first out of the bag was from Martyn Comerie in Nottingham. Well done, Martyn!

Questions & answers

Q **I have purchased Cakewalk Pro Audio 7 and I'm having a lot of fun with the audio effects. The problem I have is working out delay times in milliseconds to match the tempo of the music. Is there a formula I can use?**

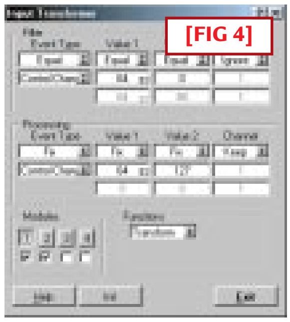
DAVID SAUNDERS

a *There certainly is: a quarter-note delay (crotchet)=60/tempo. Try using multiples and divisions of this, for rhythmic variation. For example, dividing the result by four will give you a semi-quaver delay. Multiply this further, by three, to produce a dotted quaver delay which is useful for semi-quaver-based synth patterns. For stereo delay, simply double or triple the delay time for the opposite side.*

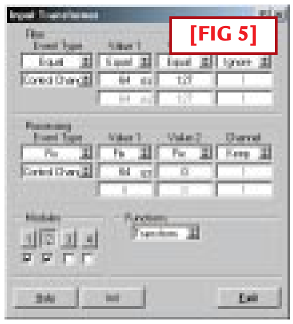
Q **Having just upgraded my MIDI controller to a seven-octave piano-weighted keyboard, my sustain pedal no longer works properly with my sound card (a Turtle Beach Pinnacle). It now works in reverse, in fact, and is active when the pedal is released rather than pressed. This makes it impossible to use. Is there a MIDI fix for this, or will I need to buy a new pedal?**

PAUL CUSICK

[FIG 4]



[FIG 5]



a *I have come across this problem myself. There are two options. Depending on the model, there may be a polarity switch inside the pedal, which will solve the problem easily. If not, there is a "fudge" you can implement if you use Cubase: using the Input Transformer, you can convert sustain pedal on events (CC64:127) to sustain pedal off events (CC64:0) and vice-versa. Figs 4 & 5 show the settings you need.*

FREE PLUG-INS

Collecting plug-ins can be an expensive hobby, especially when some of the better ones can set you back up to £300 apiece. In fact, some of the best are available for free. As I write, there are already over 80 free plug-ins, many of which can be downloaded from the Cubase web ring. Three in particular have caught my attention this month: Vellocet's Reorder [Fig 1], Prosoniq's NorthPole [Fig 2] and KwikKomp 2 [Fig 3] from Synchromesh.

For the current list of Cubase web-ring sites, visit www.webring.org/cgi-bin/webring.

◄FIG 1 REORDER SPLICES DRUM LOOPS INTO 16 PARTS AND RECYCLES THEM TO CREATE NEW GROOVES. YOU CAN REVERSE ANY PART AND CROSS-FADE BETWEEN SPLICES FOR GLITCH-FREE PLAYBACK

▲FIG 2 NORTHPOLE IS A FOUR-POLE ANALOGUE FILTER WITH LOW- AND BAND-PASS SETTINGS, ADJUSTABLE RESONANCE, AN ENVELOPE FOLLOWER AND BUILT-IN DELAY

►FIG 3 KWIKKOMP IS AN ANALOGUE-STYLE COMPRESSOR OFFERING RATIO, THRESHOLD, ATTACK, DECAY AND GAIN MAKE-UP PARAMETERS. IT'S EASY TO USE AND CAN ADD A LOT OF PUNCH TO INDIVIDUAL DRUM SOUNDS

input. For the best possible recording, ensure the synth's output and MIDI volume are set to maximum, as this will improve the signal-to-noise ratio. If you're running low on audio tracks, group like-minded instruments together.

We'll look at the processing options in next month's *Hands On Sound*. In the meantime, in the panel above are details of plug-ins you can get for free.

PCW CONTACTS

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Christmas chords

Here's something to keep you amused over the festive season. Steven Helstrip presents a selection of the finest shareware virtual studios for making music with your PC.

Over the past year we have seen some exciting developments in sound technology, perhaps most notably the virtual studio. Cubase VST certainly started something big; within 12 months of its release there were around 100 virtual effects which could connect to your virtual sound studio with virtual leads. Then came the wave of virtual synthesisers and virtual samplers. It came as no big surprise that these could connect to your virtual studio with virtual MIDI interfaces.

Five years from now, professional studios will exist purely to record orchestral, folk and rock music. Pop, dance and the remaining electronic genres will be recorded at home or, at best, in project studios driven by software. There simply won't be the need for expensive DSP and valve hardware; it will all be modelled in software.

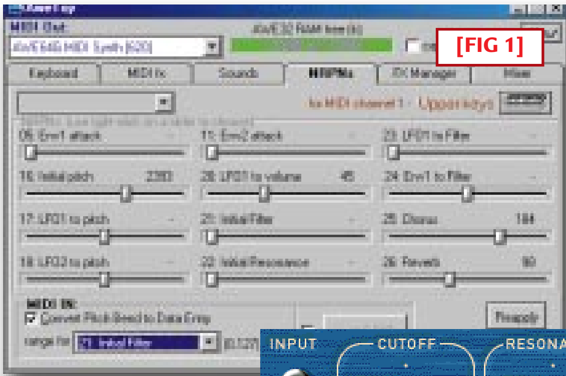
Even the sound engineer is under threat. On page 286 we take a look at a new plug-in from Steinberg that can learn the sonic characteristics of a platinum-selling album and apply it to your own bedroom recordings. Whatever next?

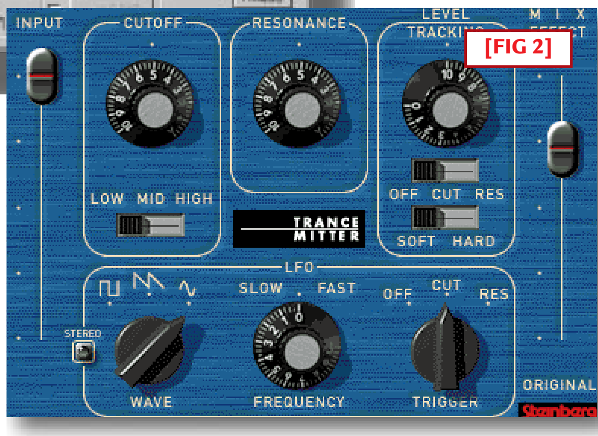
Festive fun

As it is coming up to Christmas, we are going to look at the five best shareware virtual studios for making music with your PC. It's a mixed bag of synths, samplers and utilities so there should be something here to keep you entertained over the festive season.

AWE Toy

If you've got an AWE sound card, you can't afford to be without AWE Toy [Fig 1]. It provides access to all the card's synth, SoundFont and audio parameters via nine panels. MIDI processors include


[FIG 1]



[FIG 2]

an arpeggiator, pseudo echo and a chord generator, all of which can be routed to your sequencer to record performance data in real time. There's support for virtual SoundFonts and setups can be saved on a per-song basis. Available for free download at www.maz-sound.com.

Hubi's LoopBack Device

If you want to control software-based MIDI applications from your sequencer, you're going to need a virtual MIDI interface. Hubi's LoopBack Device provides up to four MIDI ins and outs.


[FIG 3]

It's a doddle to set up and rates highly as one of my most useful utilities. It's available for free download at www.maz-sound.com.

Signum 1100DX

Here's an Akai S1100 sampler in software [Fig 2]. The entire operating system has been implemented, right down to sample editing, and there's even support for VST and DirectX plug-ins. Sample banks, or programs, can be created by simply importing a wave file and assigning a MIDI channel to it. With more features than a Swiss army knife and support for multiple sound cards, we'll be seeing a lot more of this in 1999. www.signum.it/1100/1100.htm.

Trancemitter

Trancemitter [Fig 3] is a free VST filter module. What makes it interesting is that its frequency cut-off parameter can be triggered by the level of the input signal, making it similar to an auto-wah effect. There are parameters for resonance, LFO rate and level tracking. As more applications are supporting VST plugs, everyone should have a copy. www.steinberg.de.

VAZ

VAZ [Fig 4, p286] is a virtual analogue synth based on the classic Arp Odyssey. It comes with over 180 synth patches to start you off and has very low latency (typically 20-40ms). The noises that emanate from this software are massive,

Questions & answers

Q Thanks for the great Sound pages in *PCW*. I have a question regarding PC-based recording and would be grateful if you could help, as I've been given a lot of conflicting advice and need someone with a similar perspective. I'm a keyboard player looking to purchase a new PC for recording and sequencing work. I need to be able to record and simultaneously playback eight audio tracks through an Event Gina sound card. I've been given to understand (by a PC supplier) that with this level of recording, SCSI data storage is the only way to go. Is this true? The new Ultra DMA drives also offer fast data transfer. I have budgeted for the extra cost of two 4.0Gb SCSI drives and an Adaptec controller card. Can you tell me if I'm wasting my money? The additional cost is approximately £650 over an equivalent UDMA-equipped PC, which could pay for some quality mics and a pre-amp instead. I would also be using software effects on a number of tracks. The basic system is a PII400MHz with 128Mb of RAM.

GEORGE BRITTON

...two 4.0Gb SCSI drives and an Adaptec controller card. Am I wasting my money?

a *I'm happily running Cubase VST on a 266MHz PII equipped with a Gina card. With my UDMA hard disk I can playback 12 tracks simultaneously without a hint of glitching. This leaves plenty of headroom for software-based effects: I usually have a couple of compressors, a delay and a reverb on hand. SCSI drives generally outperform IDE drives, providing faster access and transfer rates. However, playback of eight audio tracks only requires 1.2Mb of data to be read from the disk per second. Under perfect conditions, UDMA drives can deliver up to 33Mb per second. If you're able to dedicate two UDMA drives to audio files, you're unlikely to encounter any problems.*

Q I use a lot of samples in my music and recently got hold of a copy of CD-DA to allow me to digitally extract audio from CDs. My problem is that whenever I put an audio CD into the drive, the disc starts to play automatically. This is quite irritating, as I have to stop and close the Media Player before I can carry on. I know there's a way to turn off this feature but I don't know how. Can you help?

ANDREW HUNDLEY

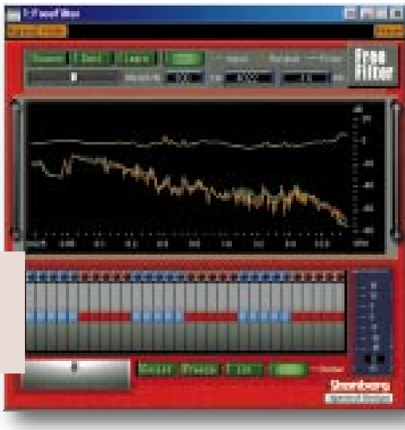
a *There are two ways around this problem. You can either hold down shift when closing the CD tray, or a more permanent solution is to disable Windows' Auto Insert Notification. To do this, open the Control Panel and load System. In the Device Manager section, select the CD-ROM setting and double-click on your drive to open the Properties dialog. Auto Insert Notification can be found in the Settings tab.*


[FIG 4]

ranging from Prophet-like pads to faithful 303 basses. You've got to try this to believe it. Available for free download at www.software-technology.com.

Steinberg FreeFilter

If you want your music to sound as good as the CDs you buy, FreeFilter could help you on your way. It's a DirectX plug-in that can learn the sound of one piece of music, then cunningly apply it to another. So, for example, you can take a song that you feel has been well mixed and EQ'ed, let FreeFilter analyse its frequency content, then adapt its EQ settings to your own mix. Put simply, it's a virtual engineer that can add a touch of sparkle to home-made recordings. The 30-band EQ can operate like any other graphic equaliser to provide up to 15dB of gain and cut. The fun doesn't start, though, until you have a two-track mix of a tune that needs a hint of spice. To analyse a platinum-selling single, all you have to do is enable the Source button and play the wav file from your audio editor. FreeFilter then creates a "fingerprint" of its frequency content [Fig 5]. Before applying the settings to the destination file, you must first allow the plug-in to learn its frequency content, which is effectively the same process.


▶ FREEFILTER, THE IMPRESSIVE PHONIC PHOTOCOPIER

Then comes the fun bit: click the Match button and allow your track to be transformed. All this happens in real time and you can set the strength of the modification from zero to 200 percent. Of course, there's a great deal more to a good mix than just EQ, but given the right source material with which to work, FreeFilter can work magic. It's a bit pricey at £299 (inc VAT) but this is a product that delivers professional results.

• Arbiter Audio is on 0181 207 5050.

PCW CONTACTS

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The MIDI-Files

The way your MIDI files are structured makes all the difference to the resulting sound. Steven Helstrip shows you how it's done.

Music that's written for General MIDI should sound roughly the same on any GM-compatible device. That is to say, with the right instruments assigned to each part, and with the effects set up in the way the composer intended. You would also expect to maintain the panoramic position and relative balance between instruments. So why is it that songs often come across differently when played back on equipment other than what it was recorded on? Or, a few months down the line, it sounds different again on the very equipment with which it was created?

▲**Fig 1** Here's how your setup bar should look. Not very interesting I know, but it's the best I could do

The bar is open

Judging by the MIDI files I come across, and from my own experience, more often than not it's because the file is not structured correctly. Whether you're writing for General MIDI instruments or a mixed bag of synths in your bedroom, it is essential to have a setup bar in your arrangement [Fig 1] to configure each instrument on every channel. So what should go in this setup bar? Firstly, to ensure you're working with a clean palette, you should insert a GM/GS reset at the start of bar 1. This is a System Exclusive command that effectively initialises the synthesiser chip to its default settings. To do this, create a new part and use the list editor to insert a System Exclusive event with the following settings:

```
F0,41,10,42,12,40,00,7F,00,F7
```

As System Exclusive events are not (MIDI) channel specific, this can go on any track. If you have an XG-compatible device such as Yamaha's DB50XG, the initialisation command is:

```
F0,43,10,4C,00,00,7E,00,F7
```

Having to do this each time you start a new song would be a tad tedious, so save the part on its own for use in future arrangements. If you work with Cubasis, which was given away free on our October issue cover CD, you might like to save this part with the default song. The default song (def.all), which can be found in the Cubasis folder, loads automatically each time the program starts. It takes approximately 50ms for these commands to be executed so eave a gap of, say, one beat before sending subsequent messages. These should include a bank number (CC:00), a program number (program change), a volume setting (CC:07) and a panoramic position (CC:10). Equally, if you have changed the effect-send levels for an instrument, controllers should also be present in the setup bar. Controllers 91 and 93 are used to configure reverb and chorus levels respectively.

> ***Reverb on the bass can clutter up the lower end of your mix, so avoid it***

To get more out of the default song, why not set up parts containing these settings with standard values for each channel? Then, each time you begin a new song, all you have to do is open the List editor and change the relevant controllers. The standard values, which are the same as those which follow a reset, are as follows: bank and program changes 0, volume 100, pan 64, reverb 40, chorus 40.

Just so you get the general idea about this process, I've carefully crafted a default song with all these settings in place. It can be found on this month's cover CD in the *Hands On* folder. Just copy it to your Cubase or Cubasis folder.

General MIDI tips

1 To create more interesting string textures, try layering two or more patches. Copy your part to a new track in your sequencer and select a new patch. To my ears, String Ensemble 1 (program change 48) works well with Synth Strings 2 (program change 51). For more variation, try transposing one of the parts up or down an octave. Panning the two instruments left and right can also make the part more interesting.

2 To add depth and contrast to your music, vary the amounts of reverb you apply to each instrument. Reverb tends to make instruments sound distant, whereas "dry" instruments sound more up-front. Reverb on the bass can clutter up the lower end of your mix, so try to avoid it.

3 If you're programming an acoustic or electric guitar, add the occasional fret noise (program change 120) on a separate track. *Keep the level to a point where it can just be heard*. If you pan the guitar to one side of the mix, don't forget to do the same for the fret noises.

4 Delay, or echo, can greatly enhance a solo instrument such as a flute. However, most GM synths and sound cards cannot produce delay and reverb simultaneously. To get around the problem, copy your solo instrument to a new track and select an unused MIDI channel. Next, simply move the part on the second track forward by an eighth, a crotchet or even a whole bar, depending on the style of music. Use either volume or expression (CC:11) to lower the volume of the delayed part and pan it slightly to the left or right.

5 When you're programming drum tracks, don't overlook the percussive instruments at the higher end of the main GM set. These include melodic toms, synth drums and a useful reverse cymbal.

Free plug-in for VST

Steinberg is giving away a new VST plug-in in return for your email address. Now you can't say fairer than that, can you? It's effectively an automatic gate, aptly named Chopper. It lets you set the tempo and rhythm of the gate and provides

▲CHOP-UP YOUR LOOPS AND VOCALS WITH STEINBERG'S FREE VST PLUG-IN

parameters for intensity and wet/dry mix. It's available for download at www.steinberg.net. Go and get it.

Questions & answers

Q I recently bought a Philips CDD 3610 CD-ROM writer and am trying to copy old cassette tracks on to CD with it. The only way I can find to do this is by using Windows 95 Sound Recorder. The sound quality is okay but there is one major and one minor problem. The latter is that the recording level has to be set manually. The former is that it records up to about 53 seconds of music and then stops dead. Do you have any advice?

PATRICK MURPHY

a *We certainly do. Sound Recorder works by recording audio into main system RAM, which explains why you're only able to record around 53 seconds. A dedicated wave editor such as Cool Edit Pro can record and edit audio files up to 1Gb by recording direct to disc. That translates to around 90 minutes of audio which is just enough to copy an entire cassette.*

Cool Edit also has an adjustable input gain to set the recording level. The shareware release can be downloaded from www.syntrillium.com/cep/.

▲COOL EDIT RECORDS TAPES, CDS, VINYL, MINIDISC, DAT... ANYTHING, IN FACT

THE ORCHESTRA GOLD EDITION

The Orchestra is the first in a line-up of new SoundFont discs from Sonido Media. With over 300Mb of strings, woodwind, brass and percussion, it's got all the essentials for writing for a wide range of orchestral styles. The Orchestra is loaded with usable patches: it's not just a bank of redundant orchestral runs, passages and crescendos that don't fit into the music you're trying to write. The instruments have been recorded and put together well, with many patches containing up to 14 samples split over as many keyboard regions. Individual SoundFonts are provided in two sizes: up to 2Mb and up to 4Mb. And for those who want to replace their GM soundset altogether,

there are 8Mb and 12Mb orchestral banks. The string sections comprise bass, cello, viola and violin all played in piano, forte, marcarto and pizzicato styles, and there are solo and ensemble patches for all sections. The quality is not consistent throughout, but given the asking price of just £29.95, you won't be disappointed. On the disc there are an additional 21 free patches from Sonido's SoundFont range, including a 4Mb grand piano, guitars, and a mixed bag of analogue patches.

★★★★

Price *£29.95*
Contact *Time + Space 01837 841100*
www.sonidomedia.com

PCW CONTACTS

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Panel beaters

Have your cake and eat it, too: Cakewalk Pro Audio users can create or modify existing panels. Steven Helstrip shows you how.

StudioWare panels are arguably Cakewalk's finest feature, enabling you to control and set up track parameters or any MIDI device that responds to CCs (Continuous Controllers) and System Exclusive or MCI commands (Media Control Interface). Panels are included for many popular MIDI devices, from GS synths through to external audio workstations. If you use Cakewalk Pro Audio, it is possible to create or modify existing panels, although to start with we're going to tackle the basics of implementing and using them. Over the past two months in this column we've designed a mixer map for Cubase to control the AWE synth parameters; but how do you go about it in Cakewalk?

- **Choose File-Open** and select StudioWare from the file-type pop-up menu; the AWE panel is the first in the list. There are two ways to assign the controllers to any track: either position the track switch, which can be a tad tricky, or right-click in the panel to open the widget property box.
- **To record** your movements (known as automation) enable the record icon in

NATIVE ESSENTIALS: X MARKS THE SPOT

RECOMMENDED PERSONAL COMPUTER WORLD

In the August column I looked at the low-cost EasyWaves bundle which included a preset-only reverb and combined EQ, compressor and gate module. TC Works' Native Essentials offers a similar line-up of three separate components: Q equalisation, R reverb, and X dynamics.The accompanying literature boasts that these are the most processor-optimised algorithms to date, so if the TC Native Reverb plug-in is anything to go by, we're in for a treat.

Q features three bands of EQ which can be configured as either high/low shelving filters, notch or parametrics with a +/- 18dB range. Perhaps the most intriguing device here is the joystick controller, which enables you to sweep through the frequency spectrum with configurable gain.

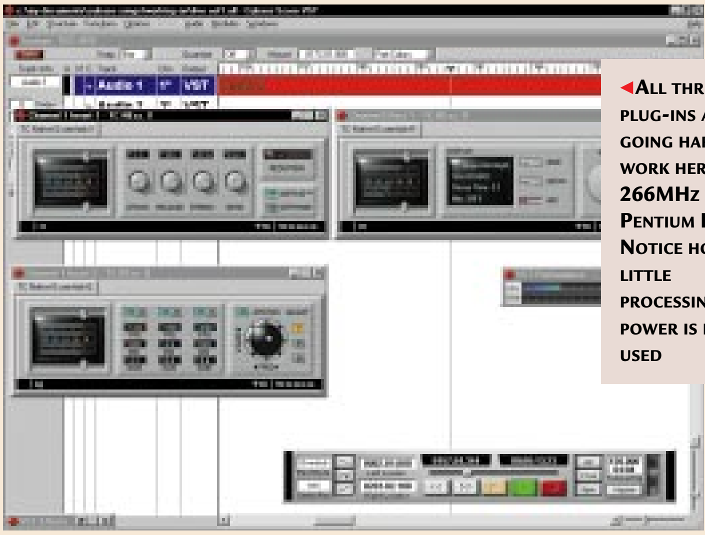
Digital EQs are generally considered to be a tad harsh, or cold, when compared to their analogue counterparts, so TC has come up with an algorithm to smooth the edges. Called SoftSat, it works by applying a soft harmonic distortion and prevents digital clipping. This enables you to boost sensitive frequencies all the way up to 18dB without experiencing undesirable clicks in your audio.

X can be configured as either a soft or hard knee compressor and features automatic gain make-up, bringing material right up to 0dB. This makes off-line normalising redundant and is a great timesaver. Parameters for attack, release, threshold and ratio are offered and SoftSat can be switched-in to provide warmer-sounding compression.

R is based on the Native Reverb package mentioned earlier and offers 18 presets with adjustable decay and mix. There's every type of reverb you could wish for, from room to cathedral ambiences with both bright and dark settings. The quality is stunning for what is effectively a £50 reverb, and long decay times sustain the source material with remarkable detail.

Native Essentials sounds as every bit as good as it looks and could become the standard plug-in suite for users on a budget. It has all the essential ingredients: high-quality effects, minimal processor overhead, flexibility, and great value for money. A demo can be downloaded from www.tcworks.de.

Price £149 (£126.80 ex VAT)
Contact Arbiter Pro Audio 0180 207 5050
★★★★★



◀ALL THREE PLUG-INS ARE GOING HARD AT WORK HERE ON A 266MHZ PENTIUM II. NOTICE HOW LITTLE PROCESSING POWER IS BEING USED

Questions & answers

This month's Q&A has one more question than I have answers for. Reader Mike Newell has a problem that has baffled the technical help department down at Et Cetera Distribution, and now myself. He writes:

Q I have a Midi-edge 1 x 4 MIDI interface card which has a Yamaha DB50XG daughterboard on it. This works superbly with both Cubase and Cakewalk to provide superb-quality sounds. However, I cannot use it for digital audio playback so I have tried installing a bog-standard SoundBlaster card. But when I do so, the Midi-edge refuses to work, claiming that whatever port and IRQ settings I use are wrong. I have tried *every possible* combination on the card, using the jumpers. Both cards work fine on their own, but not together, even though they have independent IRQs and port settings. Any ideas?

MIKE NEWELL
g1hgd@aol.com

a ***In theory there is no reason why you cannot get these cards working in the same PC and I'm convinced somebody, somewhere has already done so. After all, they're not an unusual combination. If anyone can help, or has a suggestion, please contact Mike.***

Q I am confused about sub-woofers. What are they, and is it worth getting one if I am interested in composing and listening to music on my PC?

A ATKINSON

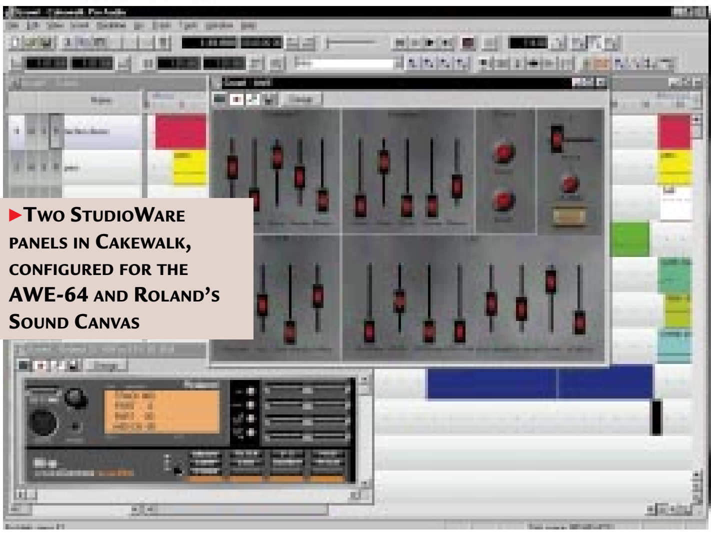
a ***A sub-woofer is a loudspeaker that is dedicated to reproducing the very lowest sounds that we can hear; anywhere from 20Hz to around 200Hz, depending on the model. Many of today's so-called multimedia speaker systems comprise a sub-woofer that is designed to sit on the floor, and two "full-range" satellite monitors which should be positioned at ear height. When producing commercial music it is important to have accurate sub-bass monitoring to get a feel for how the mix will sound in a club or arena environment, but really, investing in a sub-woofer merely for home use is sheer indulgence. However, if you need deep and powerful monitoring, which can greatly enhance video and games too, a sub-woofer could be just what you're looking for.***

Q My son uses his computer to construct music using Digital Orchestrator Pro, a Turtle Beach Tahiti sound card and the Yamaha DB50XG daughterboard. He is experimenting with the MIDI effects and, according to him, there are 50 of them. But he seems unable to find a description of what they do or how to use them. Where can he find this?

NILS ANDREAS ERSTAD

a ***There's some useful information on Yamaha's XG web page at www.yamaha.co.uk/xg/reading which explains all there is to know about accessing and using effects.***

▶VISIT YAMAHA'S XG WEB SITE FOR INFO ON EFFECTS, SPECS AND XG PRODUCTION TIPS



▶TWO STUDIOWARE PANELS IN CAKEWALK, CONFIGURED FOR THE AWE-64 AND ROLAND'S SOUND CANVAS

the panel and continue to record as normal. The icon to the right, which looks like a mini-fader, turns on the widget update to play back your movements in realtime.

☛ **It is often useful** to group widgets to allow you to control, for example, the volume of multiple tracks simultaneously. Groups are created by selecting two or more widgets with the Control key. Any widget can then be used as the "master" fader and all grouped widgets will move, relative to their start and end positions. They are defined by positioning a widget with the shift key: if the entire group is closer to the maximum value, you also set the maximum value for that widget, and vice-versa.

☛ **Crossfades** can be achieved by setting widgets off in opposite directions. First, select the two faders and position them at their mid point; create the group and move both to their highest position. Then, with the shift key, drag the second fader to its lowest position and move the first fader to *its* lowest position without shift. Finally, with the shift enabled again, drag the second fader to its highest position, and hey presto!

Piece of cake

Panels which include complex graphics can really grind slower machines to a halt, particularly when it comes to screen redraws. So, in order to disable graphics, add the following line to the [WinCake] section of the Cakewalk.ini file:

```
PanelsShowBitmaps=0
```

PCW CONTACTS

Steven Helstrip *can be contacted via the PCW editorial office (address, p10) or email* **sound@pcw.co.uk**

Big finish

Bring your instruments to life. Having created a mixer map to control parameters, Steven Helstrip shows you how to complete the assignment of controllers to the front-end of your synth.

In last month's column we set out to create a mixer map to control the AWE synth parameters. Like any other synthesiser, whether it's an inexpensive sound card or a £2,000 workstation, the AWE preset instruments fail to do the card justice. It's only when you start to program for yourself that the instruments come to life. So let's complete the front end by assigning the remaining controllers.

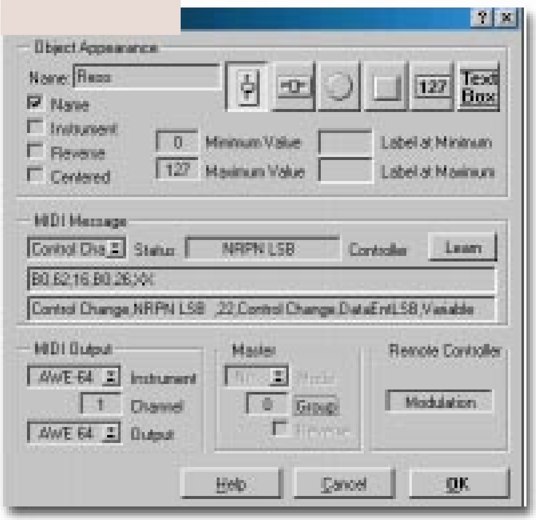
So far we have created two objects: a switch to "turn on" the NRPN MSB and a fader to control frequency cut-off, (effectively a low-pass filter). With the fader set to 127, the filter is fully open and there is no change in timbre. As you lower this value, the sound becomes more muffled as the higher frequencies are filtered out. There are nine more controllers to complete the filter section, including resonance, a six-part envelope, envelope depth and LFO depth.

Resonance

Resonance is the second most important parameter in any filter section. Essentially it allows you to route the filtered signal back into the filter circuitry, which in turn creates a feedback loop. The result is greater emphasis around the frequency range of the filter cut-off.

High settings can produce squelchy 303-like blips, although some care must be taken not to overload the feedback loop at low frequencies as this can damage your speakers at high volumes.

FIG 1 HERE'S THE OBJECT DEFINITION FOR RESONANCE



Fade to filter

Rather than create a new object for resonance, copy the filter cut-off fader by dragging the object with the Alt key. This automatically opens the object definition dialog, letting you enter the new controller values. The input line should read B0,62,16,B0,26,XX [Fig 1]. The decimal translation is:

Hex	Meaning (Decimal)
B0	CC Status Byte
62	CC 98: NRPN LSB
16	Parameter 22: Resonance
B0	CC Status Byte
26	CC 38 Data Entry LSB
XX	Variable

The filter envelope allows you to shape the filter over time and provides delay, attack, hold, decay and sustain parameters. A master control to set the overall envelope depth is also supported, along with depth for the first LFO (low-frequency oscillator).

The AWE has two LFOs which can be set independently, with parameters for rate and delay. When applied to the filter envelope, you can create anything from subtle tremolo and Leslie effects to warped, Prodigy-like synth patches.

Fig 2 shows the remaining parameter values to complete the filter section.

As with the resonance controller, copy an existing fader to create the new objects, changing the input line and names accordingly. Note that the minimum and maximum values for envelope depth should be set to 64 and 127 respectively.

To make the panel layout more authentic, I have copied the design of the Roland JP-8000 synth and assigned dials

[FIG 2]

Filter envelope parameters

Parameter	Input Line
Delay	B0,62,04,B0,26,XX
Attack	B0,62,05,B0,26,XX
Hold	B0,62,06,B0,26,XX
Decay	B0,62,07,B0,26,XX
Sustain	B0,62,08,B0,26,XX
Release	B0,62,09,B0,26,XX
Env. Depth	B0,62,18,B0,26,XX
LFO1 Depth	B0,62,17,B0,26,XX
Amplitude Envelope	
Delay	B0,62,0A,B0,26,XX
Attack	B0,62,0B,B0,26,XX
Hold	B0,62,0C,B0,26,XX
Decay	B0,62,0D,B0,26,XX
Sustain	B0,62,0E,B0,26,XX
Release	B0,62,0F,B0,26,XX
LFO1 Depth	B0,62,14,B0,26,XX
Pan	B0,0A,XX
Effects	
Reverb Send	B0,62,1A,B0,26,XX
Chorus Send	B0,62,19,B0,26,XX
LFO1	
Rate	B0,62,01,B0,26,XX
Delay	B0,62,00,B0,26,XX
LFO2	
Rate	B0,62,03,B0,26,XX
Delay	B0,62,02,B0,26,XX
Pitch	
LFO1 Depth	B0,62,11,B0,26,XX
LFO2 Depth	B0,62,12,B0,26,XX
MIDI Reset	B0,79,XX

Questions & answers

Q I have the musical acumen of a cricket stump, so your series of articles has stimulated me to dabble in computer-generated sound and composition. I have a SoundBlaster 16 and use Evolution Audio for sequencing. The FM synthesis of the first 16 GM instruments is very good. After that, however, all instruments are just variations on an organ. How do I get more realistic instrument sounds, or is this a case of getting what you pay for? I don't have a separate MIDI Mapper Applet in the Control Panel: *should* I have one?

LOCKY@GLOBALNET.CO.UK

a ***You don't have to replace your sound card if it's just higher- quality instruments you're after. The SB16 provides a feature connector that allows you to connect a WaveTable daughterboard. By far the best upgrade is Yamaha's DB50XG, which you can get hold of for around £89, or it may be worth checking out a software synthesiser. Again, Yamaha is leading the way with its S-YXG50. This offers a better spec than the DB50XG (up to 128 voices) although there is some latency when playing the synth directly. So, the bad news is that you will still have to make do with more FM organs while recording parts. For a 90-day free demo, go to www.yamaha.co.uk. The MIDI Mapper applet was a feature of Windows 3.1. Windows 95 has a similar utility, although this is found in the Multimedia Control Panel.***

▲ YAMAHA'S VIRTUAL SYNTH. WATCH OUT! IT HAS A HUGE APPETITE FOR RESOURCES

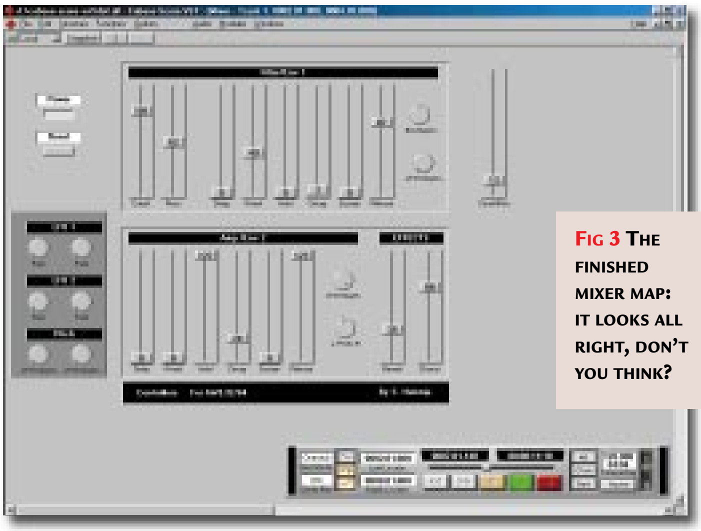


FIG 3 THE FINISHED MIXER MAP: IT LOOKS ALL RIGHT, DON'T YOU THINK?

to envelope and LFO1 depth. For the title bar, simply create a text object and choose a background colour to suit.

Envelope settings

Next come the envelope settings for amplitude, or level. This is a six-part envelope with identical parameters to the filter envelope. Using the Alt key, select the envelope objects from the filter section and drag them to a clear area on the screen. Fig 3 shows the corresponding parameter values for this envelope, along with the remaining controllers.

A standard CC:10 is used for setting the pan position. To centre the pan pot by default, check the Centred option in the object definition dialog. Continue to create new objects for each of the remaining controllers and group them under the headings: Effects, LFO1, LFO2 and Pitch. The effects parameters can be positioned beside the amplitude envelope, while the LFOs and pitch controllers are best grouped together.

To complete the mixer, all you need is a switch to reset the instrument patch in case anything should go wrong while you are tweaking away. This is a standard CC:121. However, don't forget to reinstate the NRPN MSB, or power switch, following a reset.

To create a 3D, or embossed, effect for the banks of controls, create an empty text box and select the embossed style from within the object definition dialog. When the box is correctly sized and positioned, choose Send Behind from the Mixer Local menu.

By setting up groups, it is possible to control two or more faders from just one object. This is particularly effective when the faders in question are frequency cut-off and resonance, moving in contrary motion. To set this up, create a new object (a fader, say):

1. Within the Master section, set the mode to Prop and select Group 1.
2. Open the cut-off and resonance objects in turn and set both to group 1.
3. To achieve contrary motion, select the Reverse option from one of the objects.
4. To try it out, set both frequency cut-off and resonance to 64 and "play" the new controller.

And there we go: a virtual front-end for the often untouched AWE synth. The final mixer is on this month's cover disc.

Storing with SnapShot

The intended purpose for mixer maps is to control various parameters in real time and record your movements to a special mixer track. However, using the SnapShot feature, you can store your instrument settings, or patches, and play them back at the start of an arrangement.

Once you have the desired settings in place, select all the objects (Ctrl-A) and click on SnapShot. Up to 22 settings can be stored. To "play" them back, simply click on the newly-created icon.

PCW CONTACTS

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Break-out

You wouldn't notice, but inside every sound card is a sophisticated synth just waiting to escape: Steven Helstrip shows you how to use non-standard controls to unleash that power.

Onboard every sound card there's a sophisticated programmable synthesiser just waiting to be tweaked. But with no front panels nor sliders to speak of, much of what's on offer goes unnoticed. If your WaveTable card is GS or XG-compatible, the main parameters can be edited with standard CCs (Continuous Controllers) from your sequencer, or editing software. Although this provides some access, there's nothing like getting your hands on real knobs (*no smirking!*) to fine tune, say, the filter cutoff and resonance of a bass patch. Help is now at hand, though, in the form of Phat Boy so see the box, on page 264, for my mini-review of this product.

Meanwhile, I mentioned that it is possible to control GS and XG parameters with standard CCs (see Table 1). However, if you have an AWE-32 or 64, this is slightly trickier because the cards make use of an obscure set of NRPNs (Non-Registered Parameter Numbers) for basic editing.

What are NRPNs?

Non Registered Parameter Numbers are a bunch of non-standard controls for editing MIDI parameters. Although these are effectively CC messages, a group of three or four events are needed to change a parameter value.

First, the required parameter number has to be written to CC98 and CC99 (NRPN LSB and NRPN MSB), followed by two data entry values to CC38 and CC06 (Data Entry LSB and Data Entry MSB).

This system provides 14-bit resolution, as opposed to MIDI's standard seven bits, enabling up to 16,384 controllers to be set with a value of equal resolution. This is way over the top for standard programming requirements and makes programming itself more difficult than it needs to be.

Table 1 — Controllers

GS & XG Common Synth Parameter CCs

CC	Parameter
73	Envelope Attack
72	Envelope Release
65	Portamento On/Off (0=Off 127=On)
5	Portamento Time
71	Filter Cutoff
74	Resonance
91	Reverb Send Level
93	Chorus Send Level
7	Main Volume
11	Expression (secondary volume)
10	Pan Position

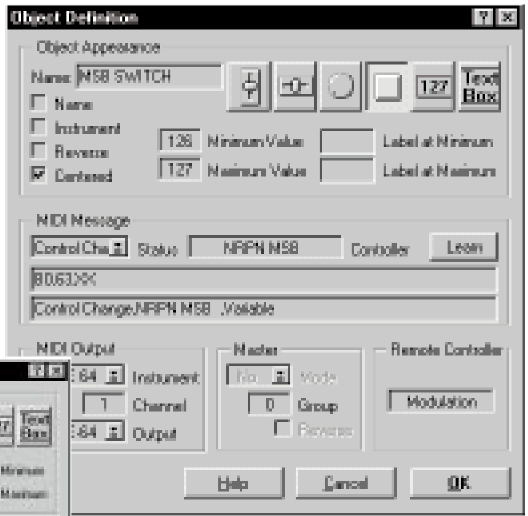


Fig 1 (above) Switch to set NRPN MSB

Fig 2 (left) Settings for the filter cut-off fader

Create a mixer map

You don't need to fully grasp NRPNs to put them to use, though. If you use Cubase or Cakewalk it is possible to create a mixer map to assign the controllers to single faders. And, over my next two columns, this is exactly what we're going to do, starting with Cubase, to provide a usable front end for the AWE cards.

1. On a new track, set the class to Mix Track by clicking in the C column and create an empty part (Ctrl P).
2. To open the Mixer Map window, double-click on the newly-created part. The first object we're going to create is a switch to fix NRPN MSB to 127. By doing so, we can then select any control parameter by sending a single NRPN LSB.
3. To create the new object select the New tool and click on the desktop, which will open the Object Definition dialog.
4. Select the switch, or button object and set the minimum value to 126 and the maximum value to 127 (Fig 1).
5. On the Input Line, enter B0,63,XX.

If you're not up on programming in hexadecimal (*and who is?*), this translates to MIDI as follows: B0 is the status byte for selecting a continuous controller; 63 in hex selects controller 99 (NRPN MSB) and XX is

COMPETITION

Win a home studio for your PC!

Here's your chance to win a Terratec EWS64S sound card complete with a full-size, four-octave MIDI controller keyboard to get the most out of it. Together they're worth around £290, but they could be yours, free, if you enter our competition.

The EWS64 S walked off with the *Editor's Choice* award in our July issue sound card group test. It offers a winning combination of hardware features, software support and value for money.

- There's a massive 64-voice synth and sampler, hardware mixing for digital audio applications and a four-channel output for Direct3D games.
- Inside the flight case there's also a CD crammed with audio utilities and applications, including Cubasis AV for combined audio and MIDI sequencing.
- No studio would be complete without a MIDI keyboard and this is no exception. The MIDI Master Pro provides a velocity-sensitive four-octave keyboard with aftertouch and a connector for a sustain pedal.
- The onboard LCD display enables you to configure the data slider to send a range of controllers, including reverb and chorus levels and there's full access to bank change parameters. There are also pitch bend and modulation wheels and buttons to transpose the keyboard range +/- three octaves.

● To enter, answer the following questions:

a) How many voices does the EWS 64 have?

b) Is it true that the EWS 64 has a built-in sampler?

c) How many octaves does the MIDI Master Pro have?

Send your entries on a postcard, or on the back of a sealed envelope to: Terratec Competition (HoS), PCW, 32-34 Broadwick Street, London, W1A 2HG.

Closing date for entries is 27th August, 1998. The first correct entry drawn will win the prize. Usual PCW competition rules apply (see p289).

the variable for setting the controller value. In this case, when the button is depressed, this will be fixed to 127.

Got it? Don't be put off, you'll only have to go though this once.

6. Set the MIDI output to the AWE driver.

7. Click OK to return to the Mixer window and resize the button object so that it's at least visible. You can create a text object if you wish to label the button as, say, Power.

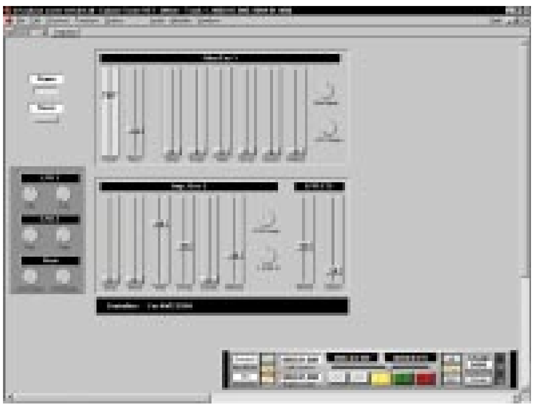


Fig 3 A glimpse of what's to come next month

Create a fader

The next object we're going to create is a fader to control frequency cut-off.

1. In the Object Definition dialog select the vertical fader and give it a name like Cutoff.

2. The Input Line (see Fig 2) needs to read B0,62,15,B0,26,XX (also see Table 2).

We now have a fader set up to control frequency cutoff. To try it out, record a part on MIDI channel 1 on the AWE output. Leave the part running in cycle mode and open the mixer map. First click the "power" button to establish the NRPN MSB and try out the fader.

Table 2 — Hex

Hex	Meaning (Decimal)
B0	CC Status Byte
62	CC 98: NRPN LSB
15	Parameter 21: Filter Cut-off
B0	CC Status Byte
26	CC 38 Data Entry LSB
XX	Variable

Next month, we'll add 25 more controllers, which will look something like Fig 3. In the meantime, don't forget to save the mixer map before closing Cubase.

p264 ➢

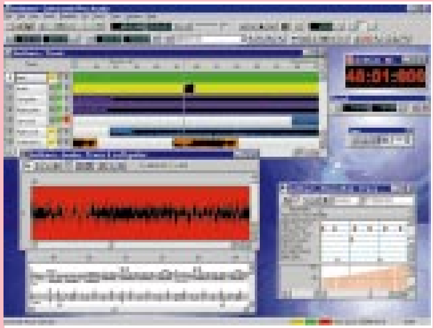
Cakewalk models

Product update

Now in version 7, Cakewalk Pro Audio is fast catching up with Cubase VST and offers up to 64 tracks for audio playback in addition to 128 real-time digital effects.

The new Console view is designed to emulate a professional mixing desk and provides access to both MIDI and audio channels, rather than having to deal with separate windows. In fact, all your arrangements can be recorded and mixed from here.

Effects can be applied as inserts, or used as auxiliary and master processors. Bundled plug-ins include a two-band parametric EQ, stereo reverb, delay, chorus and pitch shift modules. There's also support for third-party DirectX plug-ins.



Cakewalk's new and improved user interface

Automation has been introduced in the Console view and sub-mixes can be set up with colour-coded groups busses. Other high-end features include native support for Yamaha's DSP Factory and multi I/O cards.

The user interface has also been refined and the Track View now provides separate mute, solo and arming buttons per track, in addition to improved transport controls. Screen layouts can also be saved and there are new toolbars to provide quick access to the edit pages.

■ **Cakewalk Pro Audio Deluxe**
Price £369 (£314.04 ex VAT)
Contact Et Cetera Distribution 01706 228039 www.cakewalk.com

Oi!... Phat Boy

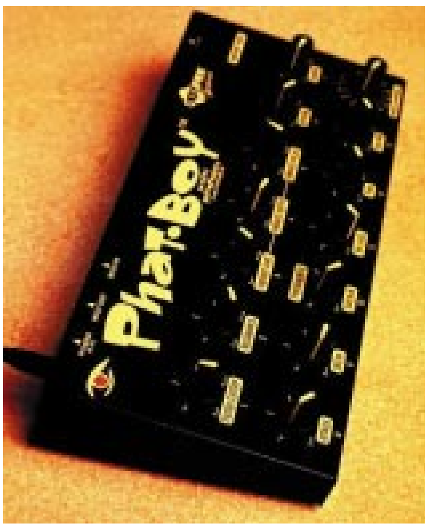
Personal Computer World Recommended

KeyFax (the same guys that developed the Twiddly Bits MIDI samples range) has come up with a solution that's so simple I'm surprised it has never been thought of before. It's an external MIDI device, called the Phat Boy, that provides 13 controllers pre-assigned to common synth parameters. It operates in three basic modes to provide compatibility with GS and XG instruments, Creative's AWE sound cards and software that has a "learn" function, such as ReBirth.

Along the top row there are controllers for filter (cut-off and resonance), vibrato (rate, depth and delay), reverb and chorus levels. From left to right along the second row there are ADSR envelope controllers (Attack, Decay, Sustain and Release), pan and volume settings. Sustain is not supported in the GS/XG specifications, so this control doubles as portamento rate instead.

As well as being able to record performance data to your sequencer in real time, the Phat Boy can also be used to generate new sounds based on preset patches. Once you have the knobs set up for the sound you're after, the snapshot button can transmit all the controllers to your sequencer so that it may be played back at a later stage. Holding the snapshot button down for three seconds performs a MIDI reset, returning the instrument patch to its default settings. A quick glance around the back reveals MIDI in and out ports, the mode select switch and the power connector. The MIDI out also doubles up as a thru, which enables you to sit the Phat Boy between your keyboard and PC without the need for a MIDI merge box. There's no on/off switch, which is really irritating, but that's the only real criticism I have. The knobs are high quality and the overall build quality is solid.

You may find it difficult to justify £160 for a box that does not itself make any sound but once you hook it up to your sound card you'll find it's worth every penny.



■ **Price** £159 (£135.32 ex VAT)
Contact KeyFax 01491 413939, www.keyfax.com
★★★★☆

Questions & Answers

Q I am looking for a PC, mainly for musical purposes. I would like to record my vinyl LPs and singles onto CD and create my own mix CDs. I have seen a PII 266MHz with 64Mb RAM and a 6.4Gb hard disk. The system comes with a 32X CD and a SoundBlaster AWE-64 card. What hardware and software will I need to undertake these tasks? I would be interested in any recommendations you could make with regards to makes and models of equipment.
David.Millis@scbrew.co.uk

A First, it's important to get hold of an audio card with a high-quality ADC (Analogue to Digital Converter). The AWE-64 is a good card but falls short in this department. The good news is that Orchid's NuSound 3D, which has an excellent ADC and high signal-to-noise ratio, comes in at £49. Secondly, invest in a high-quality cable to connect your mixer to the sound card's line input.

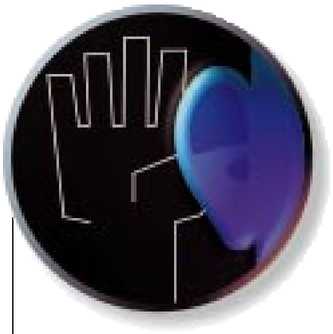
The best software option is Steinberg's Wavelab, which integrates audio recording and CD burning in one package. Version 2 has just hit the streets and now provides 13 audio plug-ins, including compression and high-quality EQs. There's also full PQ coding which will let you create track indexes to separate your mix CDs. For cleaning up vinyl scratches and surface noise levels, Sonic Foundry's Vinyl Restoration goes to work and provides excellent results with virtually no loss of dynamic range.

Wavelab uses CeQuadrat's CD-writer drivers. To check for supported devices go to www.cequadrat.com.

SCSI drives give the best performance and support audio extraction, which enables you to digitally import audio tracks from CD. Sadly I can't recommend a drive because I only know one and I understand that it has been discontinued.

PCW Contacts

Steven Helstrip can be contacted at the usual *PCW* address (p10) or via email at sound@pcw.co.uk
Orchid 01256 479898
Sonic Foundry plug-ins from SVC London 0171 923 1892
Steinberg Wavelab from Harman Audio 0181 207 5050



M people make their Points

Game, set and Match-Point: Steven Helstrip winds up his series on VST with a look at how you can use M-Points to make a groove template. Another advanced feature is automation.

Concluding our series on VST, we're going to tackle some of the more advanced audio features starting with Match-, or M-Points. We'll also take a look at a low-cost DirectX plug-in suite from Waves.

Into the groove

M-Points are markers within an audio file that map its rhythmic content by identifying where each beat occurs. With this information it is possible to quantise MIDI parts with the same feel and generate a groove template.

- To extract M-Points, first select a rhythmic pattern such as a drum or guitar loop and open the audio editor by double-clicking on the part. Ensure Dynamic Events is enabled in the View pop-up menu and select Get M-Points from the Do menu.
- To create a groove template, select Match Audio and Tempo from the Do menu, which will open the Graphic Master Track. Then select M-Points to Groove from the Audio menu, and the template will appear under Groove Quantise in the Functions menu.
- In the Arrange window, you can use the ➢

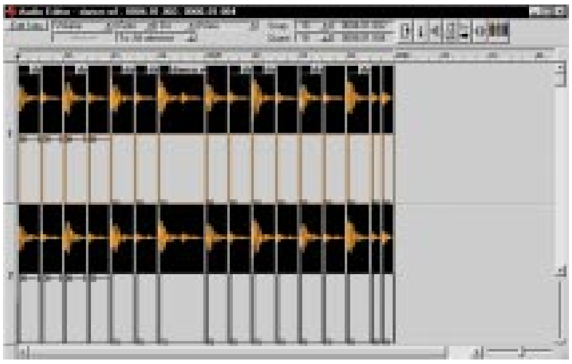


Fig 1 Chopping the loop into segments

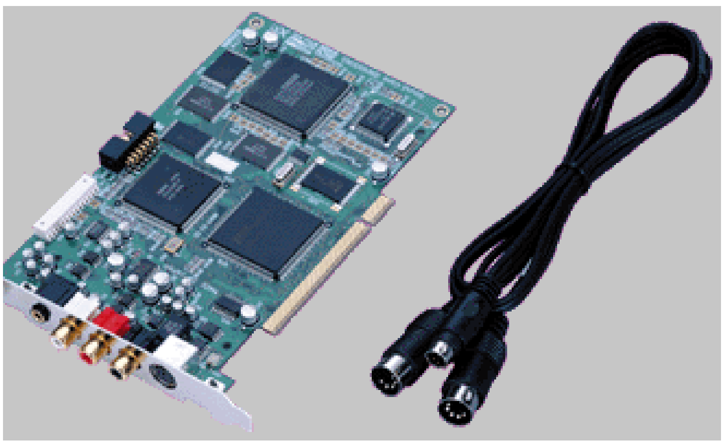
Sound developments: Yamaha SW1000XG card

Yamaha has always been streets ahead in the sound-card market and although products don't come too often, they tend to stay forever.

Nearly four years on from its release, the DB50XG is still the best WaveTable synth you can fit in a PC. But it won't be number one for much longer, because there's a new card in the pipeline that will once again take PC audio to a new level. Get ready for the SW1000XG.

With a 20Mb, 32-channel XG synth you would think Yamaha had gone slightly too far this time, but the list of features goes on. The card provides 12 channels for direct-to-disc recording, with EQ and effect inserts, seven real-time effects buses for all audio sources, a digital output and further expansion options with additional plug-in boards. But there's no mention of a sampler. Doh!

The synth is 64-voice polyphonic and has a staggering 1,267 instruments and 46 drumkits. On-board effects include 69 reverb and chorus settings and a five-band multi EQ. Since the effects are in hardware and situated on the PCI bus, there's very little CPU overhead, leaving room for DirectX software plug-ins to work alongside. All effect parameters and routing can be configured over MIDI.



As I write, there are currently three plug-in boards in development, including a vocal harmoniser and two synths: the VL-70M and the classic DX7. The harmoniser can take an analogue input from the microphone connector or be used as an effect from audio software, such as VST or Cakewalk.

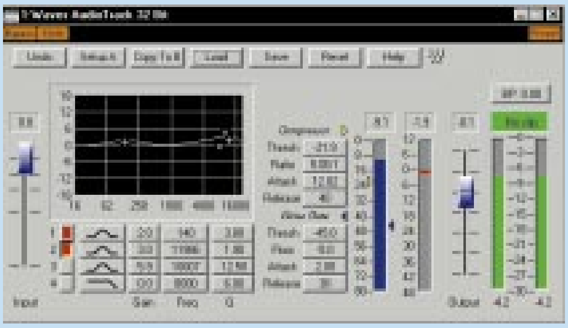
There are four types of effect on offer, including a vocoder, and harmony progressions can be played in from a MIDI track. The VL-70M card is a monophonic, physical modelling tone generator providing an extra 256 acoustic and electronic instruments with room to store 64 user presets.

The SW1000XG certainly looks impressive on paper and will no doubt be just as good when it's released in September. Shame about the sampler, though. As yet, there's no fixed price but expect it to be around the £500 mark.

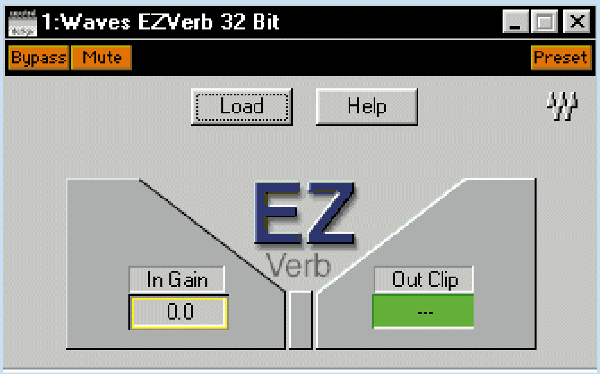
■ **Contact** Yamaha 01908 366700
www.yamaha.co.uk

Personal Computer World Recommended

Sound judgement: EasyWaves



Left Waves AudioTrack is three effects in one. It may be budget priced, but it's big on features
Below Not much to see here, I'm afraid, but there are 22 reverbs in there somewhere



EasyWaves comprises two DirectX plug-ins, EZVerb and AudioTrack, for use with DTD applications. You get Mac and PC versions on the same CD, along with demos for the entire Waves plug-in range.

EZVerb is a preset-only reverb processor based on 22 effects from the highly acclaimed TrueVerb package. Despite the lack of "tweakability" there are some really useful effects in here, from sparkling vocal plates to drenched cathedral ambiences.

Reverb for every occasion

The only parameter at your disposal is input gain, but there's a reverb to suit every occasion and, above all, it's easy to use. The quality of effects is roughly on a par with a £150 dedicated hardware unit such as the Alesis Microverb, but you'll need upwards of a 166MHz PC for glitch-less real-time playback.

The cream of this package is AudioTrack which provides an EQ, compressor/expander and noise gate in one module. The EQ section provides four independent bands with selectable hi/low shelf and bell shape filters with adjustable Q. Bands one and four can also be configured as hi- and low-pass filters.

Although this is a highly subjective area, I much prefer the sound of AudioTrack's EQ to VST's own and it's much easier to use. You can drag the handles on the equaliser graph to home in on frequencies, enabling you to work as you hear it rather than just punch in numbers.

The compressor/expander comes next in the chain and provides threshold, ratio, attack and release parameters. The level input meter is shared with the gate and two sliders enable you to visually set the threshold for both. There are no particularly special features in this department, but they do their job well and with very little processor overhead.

Perhaps the greatest advantage of plug-in modules of this kind is that they can be used on multiple channels with different settings. In a "real" studio situation you need to pay out for a compressor and gate for each audio channel, which starts to get expensive.

Highly recommended

If you only ever buy one plug-in module, I recommend EasyWaves. You get stacks of high-quality components for the price of just one noise gate. The EQ and reverbs are a valuable asset, and every studio needs a compressor at some stage.

PCW Details

EasyWaves
Price £125 (inc VAT)
Contact SCV London 0171 923 1892
www.scvlondon.co.uk
★★★★☆

p278 ➢

Q&A: recording gate patterns in Cubase

Q I have heard it is possible to record MIDI gate patterns in Cubase by playing-in a rhythm from a MIDI keyboard. I've trawled through Logical edit but cannot find a process to do this. Am I just looking in the wrong place, or have I missed something?

Neil Simmonds

Left and below

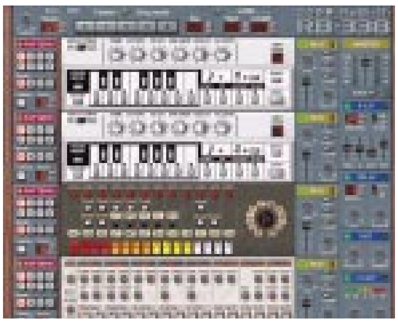
Here's how to configure Cubase's Input Transformer to record MIDI gate rhythms from a keyboard

A It is possible to record gate patterns in this way, but it depends to some extent on the keyboard you're using. It's easy enough to convert Note On information to main volume (CC7) but the problem is turning the gate off again. Some keyboards don't actually send a Note Off command when a key is released but instead send a note on with a velocity of 0, which is effectively the same thing.

Although Logical edit would naturally be the first port of call, it cannot deal with Note Off information. However, using the Input Transformer you can set up two routines to first convert velocities > 1 to CC7:127 (Gate Open) and Note Off, or velocities < 1, to CC7:0 (Gate Closed). You can then record directly to the track you want to gate. The screenshots above show the settings you need.

Coming soon — ReBirth 2.0

With the imminent release of ReBirth 2.0 there will be a direct link to Cubase, enabling individual instruments (kick, snare) to be routed to VST mixer channels. But you won't just be able to automate panning, EQ and levels, you will also be able to use VST and DirectX effects: nice touch of reverb on that snare, sir…? But perhaps better than that, the TR-909 drum machine has been added with full emulation of decay and tone settings, etc. There's also a built-in compressor and a shuffle option for each of the four components.



ReBirth gets a real 909 kit. Well, virtually...

Match Quantise tool to apply the feel of an audio part to a MIDI part. Once the M-Points are established, simply pick up the audio part and drop it on the MIDI part. If you want to quantise the other way round, the loop has first to be chopped-up into individual hits, or segments.

• In the Audio Editor, select Snip at M-Points from the Do menu and the chopping is done for you. From here you can quantise the part in any way you wish. If it all goes wrong, at least you'll have the original template stored in the Functions menu.

• With the loop chopped into segments (Fig 1, p276) it is also possible to "recycle" the constituent parts to come up with new grooves and patterns. In the audio editor simply drag the parts into any order you wish. When you have something you like, select all the parts and choose Group from the Do menu. This will enable you to copy and position the parts as a whole.

Automation in VST

Most audio parameters can be automated in VST but perhaps the most frequent are volume and pan. To accomplish smooth transitions these are best programmed in the Audio editor with Dynamic Events.

• First, select either volume or pan to be displayed below the waveform.

• Then, with the Alt key, use the pencil tool to insert gradient points. If Q and M-Points get in the way, they can be hidden by deselecting Handles from the View menu.

Copied segments will inherit the same dynamic events but this may not always be appropriate, in which case automation of levels (including EQ and effects settings) can be recorded in real-time using the on-screen faders.

• To do this, open the Mixer window and click on the Write button. When this button is lit, all movements are recorded to a special mixer track.

• To play back the automation, deselect Write and click on Read. If you make a mistake, Ctrl Z will undo the last run and you can try again.

Sudden movements

All movements are recorded to the mixer track even when Cubase isn't playing, so don't forget to disable Write after a take. This can be used to your advantage, though, since you can set up mixer states, or snapshots, to create sudden dynamic changes on one or several channels.

• To edit fader movements, select the mixer track in the arrange window and open the List editor. You can use Mask Event to view just one controller type and use the pencil tool to edit values. Events can also be deleted in the normal way.

It's a shame there isn't a dedicated editor for automation tracks since the List editor is a tad clumsy and you can't use the line tool. Inserting a string of events is also a real drag.

• Back in the arrange windows, the mixer track can be cut up and copied around just like standard MIDI tracks. If you have a complex mix on the boil, it's a good idea to make a safe copy of this, as editing out a handful of unintended controllers can set you back some time.

So there you are. I feel slightly disappointed that there isn't a new release of VST to finish off our series, but I do have news of what's to come — see the boxout on ReBirth 2.0 [this page].

PCW Contacts

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To full effect

In part III of the VST masterclass, Steven Helstrip explains VST's most powerful features — real-time plug-in effects — showing you how to use insert, mixer and master effects.

Year after year, software has made it easier for anyone to create music with their PC. Even in the early eighties we saw the Commodore 64 being used as an eight-track sequencer, drum machine and sampler.

While I and the rest of the Amiga fraternity were making the most of noisy 8-bit recordings, MIDI sequencing was gathering pace on the Atari ST thanks to Pro 12, Cubase and Creator. We now have combined MIDI and audio sequencers, and multimedia has brought about countless music packages. We've come a very long way; you only have to spend a moment tweaking the dials of ReBirth to see that. But I never anticipated the stage at which I could sit a tone-deaf friend in front of a PC and say: "Here, have a go on this while I put the kettle on", only to return to find he'd knocked out a great dance tune. The software to which I'm referring is Dance eJay (see p294).

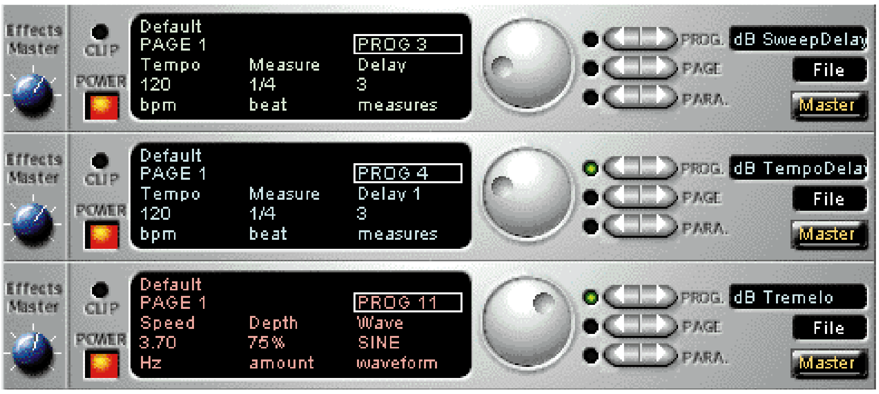

Fig 1 Dave Brown's plug-in for VST is free on the internet (see p294)

VST masterclass — part III

Amongst the most powerful features VST has to offer are its real-time plug-in effects. To kick-start your collection, VST ships with a basic range of tools that includes reverbs, delays and choruses. Since VST is entirely software driven, plug-ins rely heavily on resources, unlike sound cards which have dedicated DSPs to calculate effects algorithms. There are three basic types of effect that can be used in VST: insert, mixer and master.

1. An insert is an effect which is literally "inserted" into the signal path of an audio channel and remains dedicated to that channel — four inserts are available on each. The types of effect you would use on an insert would include compressors, noise gates, EQs and those effects where you only want to hear the return of an effect without any "dry" signal.

2. Mixer effects are on the receiving end of virtual auxiliary sends from each channel strip. Four effects can be set up and are available to each audio channel. A signal is sent to a mixer effect using the send dials in the EQ/FX window. Clicking the Pre button enables you to send a signal level independent of the channel's fader position and mute status. The return of the effect is then either sent directly to mix, or to a bus (which we'll deal with later). The most common mixer effects are reverbs and delays, as it's likely they'll be used by more than one audio channel.

3. Master effects are essentially stereo inserts on the master mix, or output. Again, compressors and EQs are likely candidates in addition to stereo enhancers.

Using mixer effects

- Select Effects from the Audio menu to select and turn on the desired effect. Most plug-ins have preset patches which are selected using the program buttons. WunderVerb, for instance, has ten effects algorithms, from large halls to gated reverbs. You can rename presets, or user settings, by double-clicking on the name.
- When routing a channel to an effect, be careful not to overload the effect otherwise clipping or distortion will occur. Bear in mind, also, that you may want to send additional channels to the effect, which will increase the input load.
- If you have a sound card with multiple outputs, or one that is capable of mixing multiple audio channels in hardware (such as the EWS64 XL), it is possible to send the return of the effect to a bus. This bus is a secondary output to mix that enables you to

New Steinberg release

3.551 has been released, which introduces Steinberg's EQ-1 EQ algorithms in the monitor section. At the press of a button, you can now toggle between standard and the new, high-quality EQ which is much cleaner and more precise with up to 24dB gain/attenuation. The driver architecture has also been optimised for better synchronisation of MIDI and audio. 3.551 is available for download from www.steinberg.net

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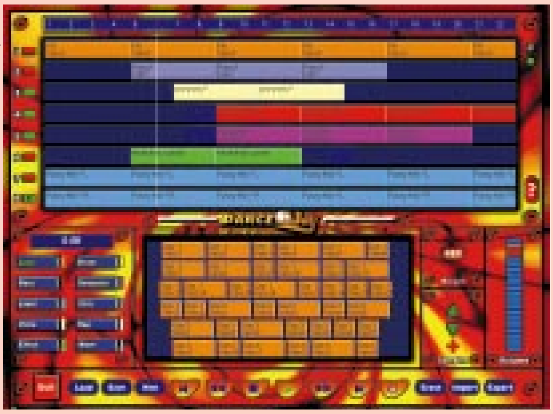
The electronic DJ



Dance eJay is an electronic DJ, although to regular readers of these pages it's best described as an eight-track sample-based sequencer.

With the techy bit out of the way, what we have here is the "Lego" version of Cubase, and it's suited to kids of all ages! The building blocks, or samples, are colour co-ordinated into various categories (loops, bass riffs, vocals, etc) and fit together in any combination of 1-, 2- or 4-bar loops. And because all the samples are in the same key and tuned to 140bpm, there really is no going wrong. Well, almost. All the loops fit to one of four chord progressions and if you try hard enough, you could layer two chords that don't work particularly well.

The arrange window enables you to drag and drop samples onto one of the eight tracks. Tracks 1-6 are mono, while tracks 7-8 make up a stereo pair. Bar numbers are indicated at

Here's a tune my mate knocked up while I wasn't looking

the top of the screen and samples automatically lock to the start of the bar. Holding down control allows you to shift sample positions by semi-quavers or sixteenths.

So what are the samples like? The overall style is definitely Euro and there are some respectable loops and synth riffs on which to lay hands. The vocals are similar in style to 2-Unlimited and, for want of a better analogy, are cheesier than a bag of Wotsits.

As for the raps, well, let's just say they're best avoided altogether. But that still leaves roughly 800 worthwhile samples with which to play around.

A full installation eats its way into 130Mb of hard disk, but that includes the full-complement of 1,350 samples. You can, of course, load up your own samples and export your opus to a wave file so that you can make a CD to send off to Pete Tong! Excellent fun and superb value. (See *Contacts*, below).

Fig 4 **Get spiced up with this valve-like distortion effect plug-in**

assign multiple channels or, in this case, effects, to a stereo fader. You can assign all your effects to the same bus for overall effect control, or to individual buses. These buses are activated in the Master section and assigned to outputs beneath each fader. Once activated, you can route effects to them from each effects rack. Note that activating buses will slow down the performance of VST.

• As the effects start to add up, your system will inevitably grind to a halt. Keep an eye on the performance indicator as this will give you some forewarning. To take the weight off your processor, the export audio option enables you to write a new file with the effects added, including any automation. This can be found in the file menu. The newly-created file will be based on un-muted audio tracks between the left and right locators. The new file will be imported to a new audio track.

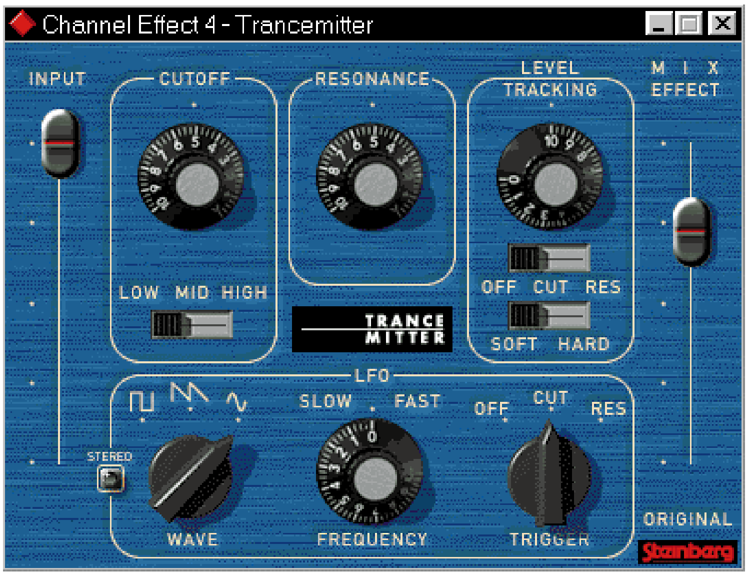


Fig 3 **Use the Trancemitter for anything which needs a touch of movement**

Get plugged in to VST on the net

You don't need to spend a fortune on third-party plug-ins for VST; there are literally dozens on the internet, for free. Check out those listed below.

• Dave's plug-ins: www.dbrown.force9.co.uk Dave Brown's plug-ins (Fig 1) consist of a tempo delay, sweeping delay and a tremolo. These delays are straight replacements for VST's delays, only they work out the delay times for you based on the tempo. The tremolo provides two independent modulators for tremolo and pan. Choose from sine, square, triangle and saw modulator waveforms.

• Trancemitter: www.steinberg.net Trancemitter is an LFO-controlled resonant filter that's superb for pads, bass-lines and drum loops. In fact, anything that needs a touch of movement. This one (Fig 3) is courtesy of Steinberg.

• Spice: www.netcologne.de/~nc-rehaagth/tr.htm Here's something to, er, spice up your kick drums and guitar riffs. Spice (Fig 4) is a basic valve-like distortion effect with just two parameters; depth and mix.

Questions & Answers

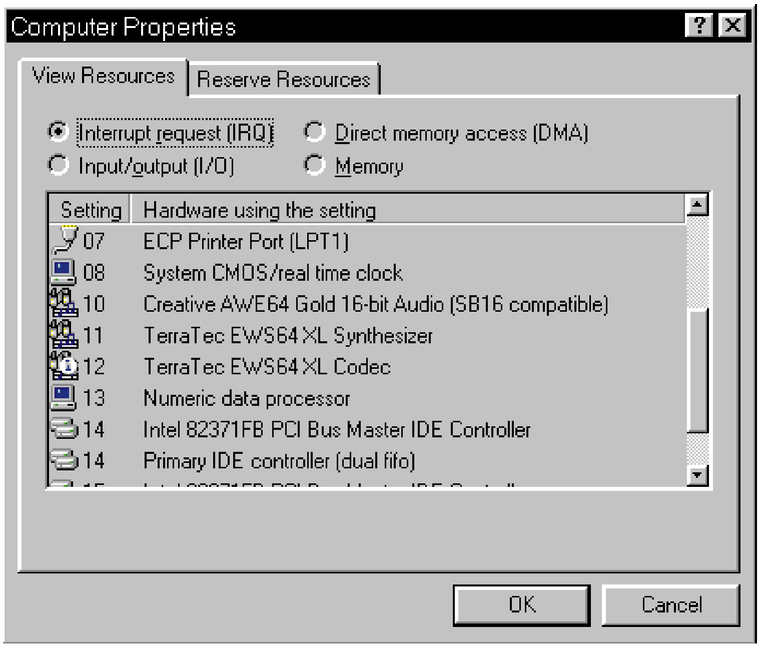


Fig 2 Under System Properties/Device Manager, click on computer to view IRQ usage

Q **A friend of mine is blind and makes his living as a semi-pro musician playing keyboards. He bought a PC with an AWE-64 (last December) and uses it with a screen-reader device called HAL95 and navigates Win95 with the keyboard. He has a WinTV card with which he can convert teletext into text files and use Creative's Text Assist to read it. He also relies on a scanner to OCR letters. HAL95 on COM2 has its own speech but it's better when used via Text Assist.**

His problems start when he wants to make good-quality recordings. The AWE-64 is positioned next to a modem, which I suspect is creating noise, but there are no free slots to which to move it, so I suggested changing the motherboard.

With his digital recording equipment my friend would like to manipulate sound in the digital domain. We know the AWE-64 has a digital output, but no input. The Maxi Sound Studio Pro has both and would seem to be a good choice. The problem is that he does not want to lose the use of Text Assist which helps him a great deal. Is it possible to have two sound cards in one machine? Can we find enough interrupts and will they conflict?

Ray Bradshaw

A I can't see any problems running two cards in the system you have mentioned, assuming you find a motherboard with more slots. I run an AWE-64 and a Terratec EWS64 XL in my PC, aswell as SCSI and MIDI interfaces. If the modem and TV cards are the only cards using interrupts, there will be ways to overcome any initial conflicts.

First, check your System settings to ascertain how many IRQs are free at the moment (see Fig 2). It's quite possible there will be only one. Depending on the second sound card you choose (check out our group test on page 206), this could well be enough. However, in the event that two IRQs are required, you can set up two hardware profiles in Windows 95: one with the parallel port enabled for use with the scanner, the other with it disabled to leave an IRQ free for sound.

The noise problem is more likely to be a combination of the graphics and TV cards, about which you can do little. However, you can remove some noise by muting the mic or line inputs (whichever is not used) in the sound card's mixer settings. Also, don't apply gain on the treble setting and avoid using 2x and 4x gain on the card's output.

PCW Contacts

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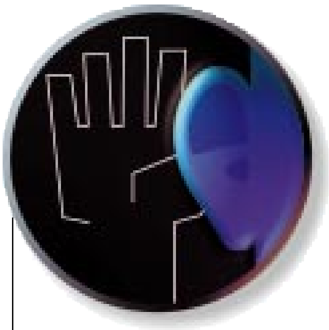
Dance eJay is £24.95 from FastTrack Software Publishing: 01923 495496 www.fasttrack.co.uk

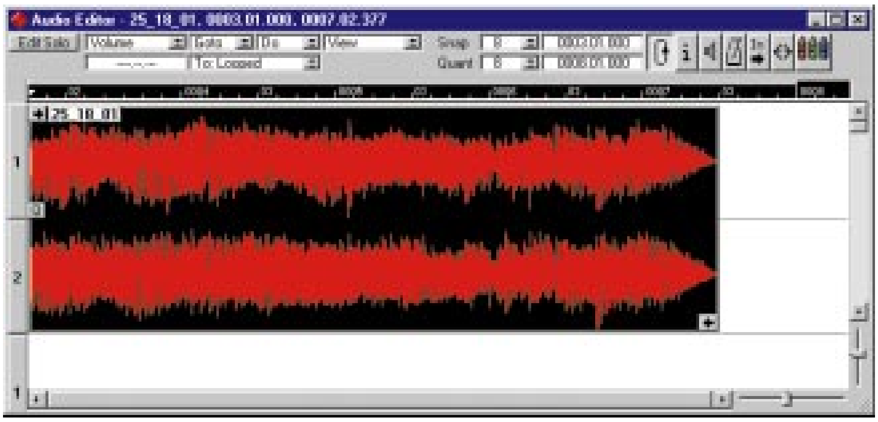


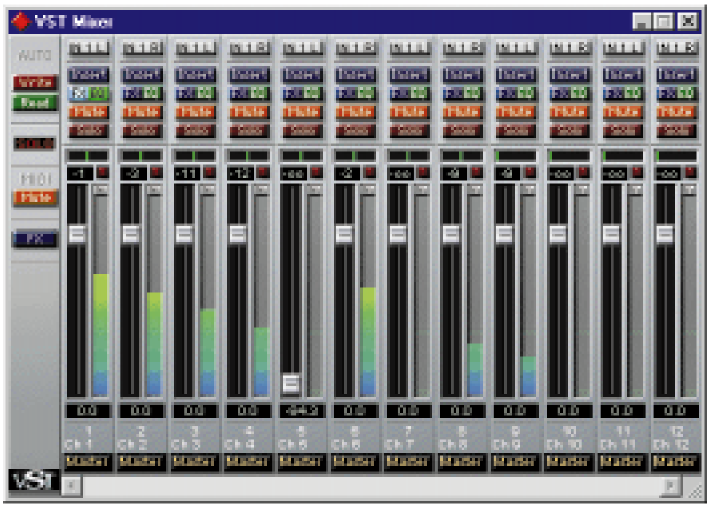
Channel hopping

How to cut a fine tune by paying attention to MIDI and audio patterns, and how to beat a path to some cool rhythms with Stomper and Hammerhead. With Ian Waugh and Steven Helstrip.

In this month's column we continue our VST masterclass and take a look at two of the coolest sound utilities of the moment which, when used together, promise to make you and your PC groove. Before we get stuck in, though, note that VST has been updated to version 3.55.

The main focus in this release is the addition of four inserts on each audio channel which, in theory at least, is cable of providing up to 32 x 4 additional effects and opens up the possibility of using plug-in compressors and third-party EQs. Support for ReCycle files has also been implemented and the Audio Engine can now be disabled at launch by holding down Shift. 3.55 is available for download from www.steinberg.net.





Above The loop in the timeline is set from bars 2 to 8, while at the current tempo, the sample runs to just over half-way through bar seven

Left The Monitor mixer window is where you mix the audio channels

Pattern cutting with VST

MIDI and audio patterns can be treated in similar ways in VST, including cutting and copying around the arrange window. With audio, the main thing to remember is the difference between audio tracks and audio channels. VST can handle up to 32 audio channels, providing you have the hardware to play back that number, but you can use any number of audio tracks in an arrangement.

Set the number of audio channels you want to use in the Audio/System menu (*see PCW, May*). Click in the Chn column of an audio track to set that track to playback on a specific audio channel. If you have selected 12 audio channels, for example, the Chn pop-up menu will list channels 1 to 12.

Stereo parts use two audio channels and must be placed on an odd-numbered channel such as 1, 3 or 5. The Chn pop-up box will then list channels in pairs such as 1+2, 3+4 and 5+6. If a track can be set to stereo, the "stereo" button in the inspector will be lit and clicking on it will toggle it between "mono" and "stereo". You can't do this if it already contains a mono recording.

You can set a track to "any" audio channel, which allows you to handle all audio channels from one track. You may want to do this if you have a system that can record on several channels at the same time. You can also set more than one track to the same audio channel, but if two parts overlap, the latest recording — the one furthest to the right in the arrangement — will "steal" the audio channel.

Time, please

One way to go about writing a song is to use pre-recorded digital audio drum and bass loops to create a backing, and add other samples and MIDI parts on top. To keep time you'll want to match Cubase's

Two of the four EQ modules available for each audio channel

tempo to the audio recording. To do this, import the file into the Audio Pool and drag it to a track so it starts at a sensible musical position. If the loop begins on a down beat, then the start of a bar is good.

Double-click on it to open the Audio Editor (note: this is not the same as the Edit Audio option in the Audio menu which is used to edit Wave files). Drag a loop in the timeline along the top of the editor for the number of bars you want the loop to play. This will be a little longer or shorter than the sample. Select the audio event (Ctrl+A) and select Fit Event to Loop Range from the Do menu. Click on Tempo and the deed is done. If the Master button is active, the new tempo is entered in the Mastertrack list.

With a song in your heart

Start a song with a tempo which fits the sample loops you're using and add MIDI parts to it. It's easier than adjusting audio to fit a tempo later on, although you can do this by selecting Audio instead of Tempo in the above example.

The Monitor (selected from the Audio menu) is a mixer which shows the audio channels with faders, pan control, solo and mute, FX/EQ and the new Insert buttons. You can adjust settings in real-time as the music plays.

No mix is complete without a dash of EQ. Click on the FX/EQ button in the Monitor, or on the FX/EQ button in the Inspector, to open the Channel Settings or FX and EQ window. You can use up to four parametric EQs per channel. Click on the arrows near the top right of the EQ window to hide and reveal the EQ modules.

Use the four Preset buttons to select a frequency range quickly. You can fine-tune the range with the Hi and Lo Limit controls. The four Q presets and the Q control determine the width of the frequency affected by the EQ: the higher the value, the narrower the frequency band. Use a narrow band to home in on a frequency and a wide band to affect a broader tonal area such as the bass or the mid range.

The Frequency control sets the centre frequency (which will be inbetween the Hi and Lo Limits), and the Gain control cuts or boosts the frequencies in this area.

EQ Tips

As a general rule, cut rather than boost. To EQ a bass guitar, use 2-4kHz for its mid range and 80-120Hz for the lower end. To make a section stand out, boost the 1-5kHz range.

Hammerhead Rhythm Station

If you're looking to get your hands on a full-blown drum machine, then have a bash on Hammerhead. It's equipped with some of the finest drum sounds from classic rhythm boxes, including the TR-909, and you can assign your own samples to any of its six audio channels. It works with loops and,

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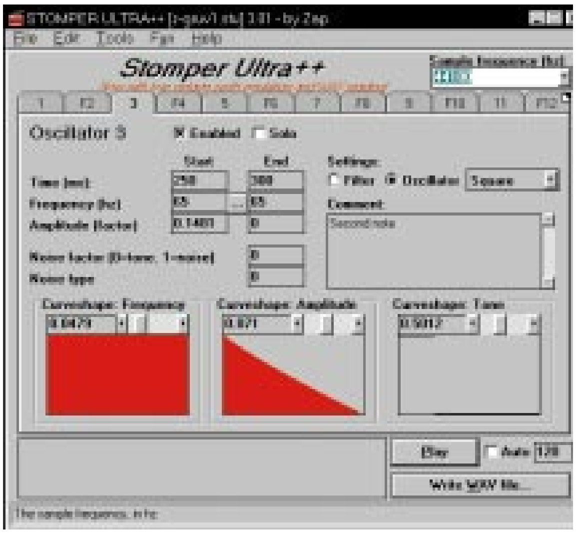
Above **Groove along with Hammerhead to create drum and percussion loops up to four bars long**



Right **Get stompin' to create lush and fat analogue-like drum and synth sounds**

unlike Rebirth, it's got distortion on every channel. It's also a breeze to program using the familiar 909 layout. What's best about Hammerhead, though, is that it's free.

To get you started there are 29 drum patches to play with, of which six are loops that will auto-tune to match the tempo of your pattern. Samples are quickly assigned to channels using a drop-down menu system and instruments can be reversed. To make your custom-built loops groove along with swing, Hammerhead has a slider to apply shuffle and the results can be heard in real-time.

Patterns can be up to eight bars in duration, although you work with one-bar loops at a time. When you've got something to try in a track, the file can be streamed to disc as a Wave file. There is currently no sync facility.

Hammerhead works with any sound card and provides a separate utility to make user sample banks. Up to six samples can be grouped, providing they are no larger than 256Kb and supplied in raw format. Lucky, then, that we also have a drum synthesiser capable of providing just that. If you check the "stretch to measure" field, loops will auto-tune to the tempo of the track. Try firing-up some bass riffs or vocal samples, too.

Stomper Ultra

Stomper uses software synthesis to produce analogue drum-like sounds. Similar to Virtual Waves, reviewed in *PCW* March, Stomper is able to generate basic waveforms and route them through resonant filters before shaping the overall sound with an envelope to simulate, say, an 808 snare.

Stomper doesn't just do drum sounds, though. It's quite possible to squeeze a respectable bass sound out of it, and for zaps and analog effects it's almost as fun and versatile as a modular synth.

Stomper can generate up to 256 oscillators and filters that can be mixed together to produce one seriously huge sound. Alternatively, oscillators can be offset to create rhythmic patterns or wave sequences. Preset waves include sine, square, sawtooth and triangle. Wave files can also be implemented and have access to the same parameters that are available for preset waves. These include start and end points for frequency (pitch) and amplitude (level). Curve shapes are available for both, and each oscillator provides a noise generator with selectable bandwidth and noise type. Tone Curve shape enables you to apply a distortion-like effect to toughen up the sound and help it stand out in a mix.

Stomper provides a useful musical frequency calculator, so if you're looking to generate a kick tuned to, say, D1, Stomper comes up with the frequency in hertz (Hz) and can copy the values to both the frequency start and end parameters. This enables you to come up with accurate glissandos so that a bass note, for example, can start on C1 and slide up to C2 for those 303-like portamento moments.

Over 30 instruments and drum sounds are provided, which conveniently double up as templates or starting points for creating your own synths. And for when you need a break, there's a built-in game of Tetris.

Stomper is musicware, which means it wont cost you anything. However, if you create a piece of music that uses a sound from Stomper, you are kindly asked to send a copy of it to the author.

● *Hammerhead and Stomper are included on this month's cover CD.*

Stomper snare

Vintage drum machines often used simple analog schematics to do their synthesis of a drum sound, implementing a ring circuit to produce a pure sine-tone with diminishing amplitude. If you add to this a touch of distortion and play around with the pitch, a wide range of percussion-like sounds can be created. When you combine a low-pitched thud with a high-pitched click, you get something along the lines of a 909 Kick. Add to this a splash of noise, pitched somewhere in-between, and you achieve something that resembles a snare.

PCW Contacts

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Further updates and info from inside.hku.nl/~bram/hammer/index.htm and stomper.base.org