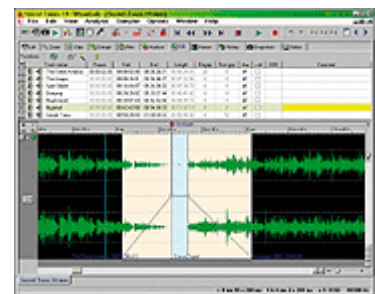




mastering masters

CD MASTERING ON YOUR PC: TOOLS & TECHNIQUES

The final stage of production — turning your mixes into a finished CD — requires some specialised tools. **Martin Walker** runs through the options and considers how best to set up and use the necessary PC software.



Lots of SOS readers have been asking me over the last few months if they can master their albums using a computer, rather than relying on external rackmount hardware to do the job.

Mastering basically involves taking the individual songs, placing them in a suitable order (which not always as easy as it sounds), adjusting their relative levels and EQ to make them sit more comfortably together, and adding any final 'fairy dust' if and when needed. Well, of course, this is possible using a computer, and once you have your recordings on a hard drive you could leave them there at every stage right up to CD burning of the final product if you wish.

However, the hot debate among traditional studio owners concerns software plug-ins, and whether or not their quality matches up to that of external rackmount outboard effects. This question is particularly important during mastering, since each make and model of the types of processor that get used, tends to have its own unique sound.

As I discussed in PC Musician SOS November 2000, there is no inherent reason why a software solution should be inferior for most types of effect, subject to it being given sufficient processing power. Indeed, some existing rackmount effects are nothing more than powerful computers in a box with additional I/O. Most modern studios seem to be using at least some plug-ins, and some have embraced them wholeheartedly as another way forward, if not the only way forward.

So, the answer is yes — you can master an album without ever leaving the comfort of your computer, and get results that are good enough to release commercially. However, mixing and mastering are very different skills, and it's important to know how to make the most of the available software tools.

Suitable Software

If all you want to do is assemble a set of already perfectly formed audio files into a chosen order and then burn an audio CD you won't need a mastering application, since you can do this with almost any CD-burning utility. However, this can be a frustrating approach, since

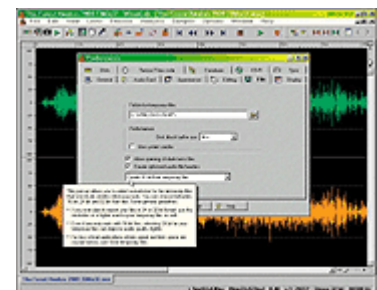
unless you can listen to the tracks in sequence before the burn, you won't be able to hear how they sound one after the other until you put the finished CD-R in your hi-fi. The latest versions of products like Adaptec's *Easy CD Creator*, Ahead Software's *Nero*, and CeQuadrat's *WinOnCD* all provide more facilities for those creating audio CDs, but there's no substitute for being able to audition and make changes to the WAV files in context.

The next step up is a list-based stereo audio editor that lets you assemble your audio files, add fades in and out where required, and audition the 'joins'. Nowadays most of these have a graphic environment that makes the process far more intuitive, as well as the ability to drag and drop each track relative to each other to adjust spacing, and in some cases even drag one across another to automatically create crossfades between tracks.

Since modern multitrack audio software is capable of running plug-ins suitable for mastering purposes, it's also perfectly possible to master in the same environment in which you record and mix. However, a dedicated mastering application may still prove easier to use in the long run, for various reasons. First, it's vital to be able to zoom in to view waveforms at single-sample level to be able to spot and remove clicks and pops, and not all multitrack applications let you do this without accessing an external audio editor. In addition, it's often easier to assemble a set of stereo mixdowns into the final order and adjust the spacing between them using a list-based approach, even if you can also view them as graphics, since dragging and dropping text in a list is far easier to deal with.

Finally, using a dedicated mastering application into which the CD-burning process is fully integrated can make the overall process even easier, especially where the final CD audio file is being calculated 'on the fly'. This is because the individual tracks always remain separate, so that you're not having to deal with single 600Mb image files for an hour-long album. In many cases the fades can also be applied 'on the fly' during the burn, which makes it easier to change things at the last moment if required, and some packages even let you apply plug-in effects to individual tracks as well.

I have come across one multitrack application that provides all these facilities — *Samplitude 2496* (formerly marketed by SEKD, but now under the banner of Magix, who also offer a dedicated stereo version called *Samplitude Master* devoted to mastering). On the PC there are several other software applications that are specifically intended for detailed work on final mono or stereo tracks. The most famous is Steinberg's *Wavelab*, now at version 3.0, which now incorporates the multitrack Montage function to assemble more complex tracks, add fades and effects 'on the fly', and has integrated CD-burning facilities.



Sonic Foundry's *CD Architect* is another elegant application, and comes bundled with their *Sound Forge Lite* editor, although most musicians will prefer the more comprehensive *Sound Forge* if it's within their budget. Sample rates of up to 96kHz have been supported for some time, but only after a wait of several years for the recent version 5.0 have 24-bit

Professional Mastering

While mastering on your PC can give good results, I certainly wouldn't claim that it gives results as good as those achieved by a professional mastering engineer. For a start, you're unlikely to have a state-of-the-art monitoring system that's flat down to 30Hz or less to accurately judge the bass end. You won't have tens of thousands of pounds worth of esoteric EQ, compression, reverb, and other goodies to tweak your sound to perfection.

Most of all, however, you're unlikely to have the same level of expertise, objectivity, and impartiality. Good mastering engineers are renowned for their 'golden ears', and their skills are acquired through years of training and experience.

files been supported as well. Syntrillium's *Cool Edit Pro* is also an excellent stereo and multitrack editing package, but doesn't have integrated CD-burning facilities. IK Multimedia's *T-Racks* provides virtual valve EQ, compression, and multi-band limiting, as well as fade options, and has recently been updated to accept 24-bit files, but unlike the others mentioned here doesn't provide graphic editing — in fact, it's more like a rackmount processor such as TC's Finalizer in approach.

Application Settings

For the best final results when mastering, the stereo audio files of each track should be at 24-bit or higher resolution. This doesn't necessarily mean that you have to record every track in your audio sequencer at 24-bit resolution, since most multitrack applications will let you mix 16-bit and 24-bit tracks at will: the important part is to make the final stereo mixdown at 24-bit, or even 32-bit if you have a suitable application like *Cubase VST/32*. You will benefit from this even when using 16-bit converters or samples on the original recordings, since as soon as multiple tracks are mixed together there will be more than 16 bits' worth of resolution anyway.

Choice of sample rate is a more thorny issue. I still use 44.1kHz since this is my target rate for burning audio CDs and although some people maintain that modern sample-rate converters are now so good that you can start at 96kHz and then down-convert at the end, I prefer not to put my audio through an extra stage of conversion. However, if you're convinced of the audible benefits of high-sample-rate recording for your type of music and gear, and your system can cope with the increased processor and hard disk requirements, go ahead, though bear in mind that 88.2kHz may be a more benign choice than 96kHz, since the down-sampling process to 44.1kHz is so much simpler.

If you're transferring a 48kHz DAT tape into your PC for mastering, you will obviously have to convert this to 44.1kHz. Since this will change the overall sound slightly, I would be inclined to do this as the first process, so that you can add further tonal tweaks to the final 44.1kHz version as required. If you have several applications capable of this conversion, try them all and compare the results, and if there are any quality options make sure that you always use the highest one — it may take considerably longer to process the whole track, but you want to lose as little quality as possible.

Whatever sample rate you choose, you should leave recordings at as high a bit depth as you can until the last moment, and then convert to the final format (normally 16-bit for audio CD burning) as the last stage, with suitable dithering. This is because any alterations you make to the audio files — including gain changes, compression, EQ, and fades — will produce rounding errors in the calculations. If they are already at 16-bit then the accumulating errors will gradually make your tracks sound coarse and grainy, and you'll

Visual Information

Many musicians find that using analytical tools helps during mastering, and I discussed many of the options in some detail in *SOS* September 2000. A spectrum analyser is useful to examine frequency response against other recordings, and can also be invaluable in spotting low-end problems that may not be audible on nearfield monitors. Most audio editors, including *Cool Edit Pro*, *Sound Forge*, and *Wavelab* now incorporate them, and shareware plug-ins are also available from Nick Whitehurst (see Contacts box). Steinberg's *FreeFilter* also has one built in, and can learn the frequency response of another track and apply it to one of yours.

A phase display can help check for mono compatibility (which is still vital if you expect radio play). Steinberg has one in its *Mastering Edition*, Nick Whitehurst incorporates one in his shareware *C_SuperStereo*, and PSP provide a *Stereo Analyser* in their *StereoPack*.

A sonogram display can help you make decisions about high-frequency enhancement, as well as spotting low-level hums, whistle and DC offset problems. Again, *Cool Edit Pro* and *Sound Forge* have one built in, while Steinberg's *Mastering Edition* has one in plug-in form.

lose fine transient detail and stereo localisation.

If your software lets you choose a resolution for temporary files, make sure this is also at a suitably high setting. In *Wavelab*, for instance, choose 'Create 32-bit float temporary files' in the File page of Preferences. While you're there, if you have separate Windows and audio drives make sure that you set the Folder for Temporary Files to the Windows one, since keeping them on a different drive from your audio ones will greatly speed up most Undo operations. You can also do this for *Sound Forge* in the Perform page of its Preferences. If you're using *Cool Edit Pro*, ticking the 'Auto-Convert all data to 32-bit upon opening' box will ensure that all subsequent editing is also carried out at 32-bit resolution.

Neat And Tidy

Once you have all your tracks as final stereo files, the first thing to do is to listen to them carefully for any basic imperfections such as clicks, pops, hums, and so on. These should really have been dealt with at the multitrack recording stage, but sometimes they slip through unnoticed. If you hear any such problems, and you still have the multitrack files, you will nearly always get better results removing the offending sounds there than on the final mix, but even with a final stereo mix there's a lot that can be done. Restoration is a complex subject, but here are a few pointers that may help.

Very short clicks and pops a few samples long can usually be neatly removed using a waveform Draw tool, while longer ones are better dealt with by a dedicated algorithm that has been designed to recognise and eradicate them. *Wavelab* has a variety of these built in to its Waveform Restorer, *Cool Edit Pro* also provides a variety of Noise Reduction options in its Transform menu, and *Sound Forge* has glitch detection available in its Find tool. The beauty of these functions is that they only alter the waveform for the duration of the click, which nearly always gives an inaudible result in the final track.

Hums are more problematic. A narrow notch filter set to 50Hz or 60Hz, depending on your local mains frequency, will get rid of the fundamental, but you will nearly always need to filter out the 100Hz and 150Hz (or 120Hz and 180Hz) harmonics as well, and this can start to compromise the overall track quality if you're not very careful. Several plug-ins, such as TC Works' *Native EQ Parametric*, Waves' *Q10-Paragrophic* and *Renaissance EQ* have suitable presets, and again, treating individual tracks rather than the final stereo mix will probably produce the best results.

The only really effective way to remove hiss is to use a specialised real-time plug-in like Steinberg's *DeNoiser*, or one that analyses a noiseprint, such as Sonic Foundry's *Noise Reduction* or Arboretum's *Restoration-NR*. The latter type will remove any continuous background noise, and can therefore be effective for hums and air-conditioning noises as well. In many cases you may get away with treating only the first and last few seconds of the track, where background noise may be more noticeable. This avoids having to pass your entire track through the plug-in, but watch out for clicks when you drop the process in and out. Many low-cost CD-burning applications have restoration tools built-in, but few of



these are suitable for professional work — use your ears before you decide whether or not to entrust your tracks to them.

Having dealt with any imperfections in the recording, it's time to top and tail your tracks, removing any extraneous material before the first note and after the final one. I normally leave a few milliseconds before the first note and apply a very short fade-in, so that there's no possibility of a click. However, be very wary of chopping off the end of reverb tails at the end of the track.

The next stage is to assemble the tracks into a suitable order — which, contrary to many people's perceptions, isn't as easy as it might sound!

Equalisation

Hopefully, your tracks will already sound fairly close to the desired result in terms of EQ. If not, then perhaps you need to spend a bit more time at the multitrack stage, since you are likely to get better results overall by equalising each instrumental track separately. Mastering EQ should be a matter of making subtle changes such that different tracks on an album fit better with one another. For this you don't normally tend to need a multi-band graphic EQ, or a complex parametric EQ (or at least not using high 'Q' settings). What you need are gentle slopes allowing you to apply moderate, wide-ranging boost or cut. These not only sound more natural to the ear, but will be far more effective at warming up or cooling down a track than lopping off or boosting isolated chunks of the frequency response.

There are loads of EQ plug-ins on the market, and they all sound different, so if you already have several to choose from you should use your ears to decide which are best for mastering. Try setting up similar settings on each and switch between them to see which sounds the most musical. Personally, I like Steinberg's *TL Audio EQ1*, since it has a warm, open sound based on TL Audio's valve equaliser and is extremely easy to use, and the 7-band parametric and switchable slope design of Steinberg's *Q-Metric*, which can process at double the normal sampling rate for an smooth, open 'analogue' sound. Another great candidate for mastering is the Waves *Renaissance EQ* (part of the *Renaissance* bundle), which has great-sounding 48-bit processing and unusual 'parametric shelf' options. The TC Works *Parametric EQ* (either the *Native* version or the cheaper *Native Essentials* one) is good for mastering as well, since it provides the option of shallow slopes right down to 3dB/octave.

Fairy Dust

Even tracks recorded in the best studios can sometimes benefit from a little enhancement. Normally this concentrates on the two extremes of the frequency spectrum, to hone and polish the bass end and add some missing sparkle at the top. There is a difference between these techniques and EQ, since they often work dynamically rather than simply raising or lowering the frequencies in question, and in some cases actually generate missing harmonics based on what's

Hints & Tips

- Make sure you regularly switch between the track you're working on and other reference tracks as a comparison, preferably including some commercial ones that have had the benefits of no-expense-spared mastering expertise.
- Every audio process costs something in sound quality, so don't put your tracks through any unnecessary stages.
- Wait at least a day after mixdown before starting to master — the two processes are very different, and mastering needs a clear head.
- Don't use a noise gate plug-in to remove low-level sounds between tracks — after all, you've got a large-screen editing environment and can add suitable fades and digital silence wherever you need it.

already there. However, the different types have already been covered in some depth in the pages of *SOS*, so let's concentrate on what's available in plug-in form for the PC.

At the high-frequency end, one of the first enhancers was Steinberg's *Spectralizer*, now available as part of their *Mastering Edition* bundle. This adds second- and third-harmonics above a chosen frequency, and like most other such plug-ins can also help enhance transient detail and stereo imaging. Arboretum's *Harmonic Exciter*, from the *Hyperprism DX* pack, works in a similar way, while PSP's *MixTreble* also offers high-frequency stereo widening and companding to further enhance the mix.

- Play back the final tracks to friends, be they musicians or otherwise, and try to keep quiet when they make comments — you'll learn more that way.

You may also want to try adding a subtle amount of virtual 'tape and tube' modelling. I discussed suitable plug-ins in depth in *PC Musician* October 2000: some to try include Steinberg's *TrueTape* and *Magneto*, PSP's *MixSaturator*, and AIPL's *WarmTone*. If you want to widen or otherwise alter the stereo image, Waves' *S1 StereoImager* and PSP's *StereoEnhancer* work very well, as does the freeware *C_SuperStereo*. It's very tempting to add too much enhancement and stereo treatment, so make sure you A/B with commercial mixes so that you don't overdo things.

The bottom end is probably the most difficult to tweak unless you have good acoustics in your studio, and even then you still won't know what's going on below the cutoff point of your monitors. Again, referring to known commercial mixes will help keep your bottom in perspective, and in many cases you could roll off everything below about 30Hz with a suitable filter to prevent unnecessary cone flap while leaving the bass almost unchanged. Suitable plug-ins to try for bass tweaks include Waves' *MaxBass*, Arboretum's *Bass Maximiser*, and PSP's *MixBass*, all of which generate harmonics based on the original bass end to reinforce it, while offering bass compression to increase the impression of bass without unduly increasing overall level.

Monitor Levels

Although it's extremely important to reference your own music to the cream of commercial releases in a similar style, it's also important to try to stick to a similar listening level whenever you are mastering, to avoid the Fletcher-Munson effect. The frequency response of the human ear varies with level, so if you play music louder it will sound more bassy and topky. If you listen at different levels each time you work you run the risk of adding too much or too little bass and high end. This is why most mastering engineers use a pink noise test signal to set up a specific sound pressure level of about 80dB SPL through their monitors at the listening position using a C-weighted level meter — Studiospares and Tandy (Radio Shack) have cheap models — and then leaving the monitor amplifier controls well alone.

A Question Of Balance

A lot of musicians normalise all their final stereo mix files, raising the level so that the loudest peaks are all exactly at 0dBFS (full scale). Now I can see the attraction of immediately making them louder, but this is totally the wrong thing to do at this stage. First of all, you make it extremely difficult to add any further treatment such as EQ or top-end enhancement, since these will both tend to raise peak levels even further. Since you're likely to want to change the overall level later on, it's also an unnecessary set of calculations to put your audio through, but most important, it won't adjust the levels so that your tracks sit well together.

Perceived loudness is a function of average (RMS) level rather than peak level, and is far better measured by a VU (Volume Unit) meter than the peak meters found on most small hardware mixers and in nearly all software. Of course, making sure that the peak level of

your recordings never reaches clipping point is vital during the recording process, but this isn't what you need for the mastering stage.

If your software can measure the average RMS level of a track then you can use this to provide an indication of relative levels between them. *Sound Forge* and *Wavelab* both provide this function, but remember that unless your music is all at the same tempo and in the same mood, some tracks will probably need to be quieter than others anyway, so use your ears as well. A ballad won't sit properly in the middle of a rock album if it's at the same level as the rockier tracks, for instance.



There are many ways to start levelling, but one of the easiest is to take the track that you feel should be the loudest, and start with that. Unless you've already compressed the life out of it, you'll probably need to increase its average level slightly to compare with most other CDs on the market. At the mastering stage, the majority of tracks can simply be treated with a look-ahead limiter like Waves' *L1 Ultramaximiser*, Steinberg's *Loudness Maximizer*, or the *L* limiter/maximiser of TC Works' *Native Bundle 2*. These can raise levels by four or five dB above normalisation levels in a virtually transparent way, by limiting only those peaks that would otherwise be clipped.

A discussion of compression options during mastering would need a dedicated article in itself, and on many tracks I'm unconvinced that that mastering compression is needed unless there's something basically wrong with the track dynamics or de-essing is required. Even then, if you've already got your multitrack sequencer sitting in the same PC then it will probably be easier to deal with there than at the mastering stage. If you do use mastering compression, a soft-knee algorithm generally sounds better, and there are many plug-in options such as PSP's *MixPressor*, TC's *Native DeX*, and Waves' *C1* and *RCL*.

Multi-band compression can help where a particular part of the spectrum needs treatment, and generally results in less audible side-effects for a given amount of gain reduction. You can use it to tweak the level of specific frequency bands without unduly altering the rest of the mix. Two plug-ins of note here are Steinberg's *ME Compressor* and Waves' *C4*, both of which offer extremely flexible click-and-drag control over band crossover frequencies.

All Of A Dither


The final stage in the mastering process is to reduce the bit depth from 24-bit or 32-bit down to the 16-bit required for an audio CD. At the same time you need to add dither noise during the process to retain as much as possible of the low-level detail from the original files. Most musicians understand the reason for dither in principle, but here's a way to make the benefits more obvious so that you can hear the results for yourself. Those who have a choice of several dithering algorithms can also use this technique to compare them.

Start by taking one of your 24-bit or 32-bit tracks and apply a fade-out with no dither to the last 30 seconds or so, leaving you with a 24-bit or 32-bit file. Now select the final few seconds of this fade, starting at a level of about -50dB or so (check the peak level in your software meters), set it to loop, and then play it back. You will have to turn up your mixer or amplifier level considerably to hear the result, and to prevent nasty damage to either your ears, amplifier, headphones, or speakers, the safest thing is to cut and paste this section into a new file to avoid accidents.

First of all, play the fade back with its normal bit resolution, and hear how it fades smoothly to silence. Now try playing back the file with simple truncation to 16-bit resolution. Plug-ins

like Waves' *L1 Ultramaximiser* let you set the Quantise value to 16-bit (with all other dither options set to None) or if you're using *Wavelab* you can change the Preferred Playback Resolution to 16-bit in the Audio Card page of its Preferences section and restart playback. You will hear that the final part of the fade will have lots of distortion as the lowest bit flips on and off.

Now you can try adding various flavours of dither noise, either using the facilities of your mastering application or using a plug-in. Set the dither to 16 bits while still playing back with 16-bit resolution. With basic dither noise added you will hear a steady hiss, but with the music now clearly audible all the way to the end of the fade. If you change to noise-shaped dither, the hiss will be less audible (since this technique locates as much of the noise as possible above 15kHz, where it is less audible to human ears).

Effectively, you are trading off a few dB of extra noise for the complete absence of low-level distortion, giving an apparent resolution of up to 19 bits in the final 16-bit file. The only thing to beware of is that some 'ultra' shape settings should only be used at the very final stage of mastering, since they add such a high peak of high-frequency noise that any subsequent editing followed by further dithering may theoretically cause distortion. However, once you understand dithering you'll retain the best possible sound quality when you finally burn your tracks onto audio CD. 

Contacts

AIPL (*Warmtone*)

www.aipl.com

Arboretum (*Hyperprism DX bundle, Restoration-NR*)

www.arboretum.com

IK Multimedia (*T-Racks 24*)

www.t-racks.com

Magix (*Samplitude*)

www.magix.com

PSP (*StereoPack, MixBass, MixTreble, MixSaturator*)

www.psp-audioware.com

Sonic Foundry (*Sound Forge, Noise Reduction 2, Vegas Pro*)

www.sonicfoundry.com

Steinberg (*DeNoiser, Magneto, Mastering Edition, Q-Metric EQ, TL Audio EQ1, Wavelab*)

www.steinberg.net

Syntrillium (*Cool Edit Pro*)

www.syntrillium.com

TC Works (*Native Essentials, Native Bundle*)

www.tcworks.de

Waves (*Native Power Pack, Renaissance Collection*)

www.waves.com

Nick Whitehurst (*C_FFT, C_SuperStereo, FFT*)

<http://ourworld.compuserve.com/homepages/NickWhitehurst/>

Glossary

http://www.sospubs.co.uk/sos/regular_htm/glossary.htm

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