

Q&A FOCUS

Recording

YOUR QUESTIONS ANSWERED



This month we're lending you our musical ears and sounding out your recording misdemeanours...



▲ It may not be pretty, and it won't work in real-time, but Nuendo offers a convolution reverb in the form of Acoustic Stamp

Bedroom bother

Q What's the best way to make your recordings sound less like they've been recorded in a bedroom... when they've been recorded in a bedroom?

Brendan

A If you're referring to the sound of the space itself, it's worth noting that many professional studios go to great lengths and expense to provide a sonically 'dead' environment so they can add a sense of space via artificial means. Conversely, many studios pride themselves on their

live rooms. If you're trapped in a bedroom, you'll have to settle for some approximation of the former. As you probably know, spaces can be artificially simulated via reverberation. Convolution – or 'impulse' – reverbs are very good at providing a simulated environment. You can find an excellent freeware convolution reverb, called SIR, at www.knufinke.de/sir/index_en.html.

If you'd prefer to create an acoustically 'live' space within the confines of your room, try using reflective surfaces such as Masonite. Anything smooth and flat will liven up the place.

If, however, you're looking for

the difference between the overall qualities of a home recording versus professionally produced tracks, there are really only two things that separate them: the quality of the gear and the skill of the engineer. Get the best gear you can afford. It's better to spend your money on one or two essential high-end pieces than it is to load your room with mediocre bargain-bin knockoffs.

Contrary to what some manufacturers' glossy ads proclaim, all the gear in the world won't improve your recordings if you don't have the requisite engineering chops. However, some things only come with experience and trial-

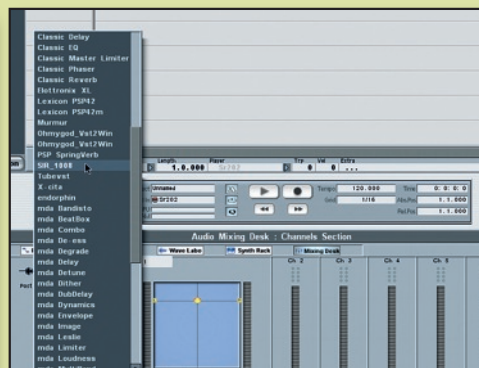
and-error. Microphone placement is a case in point: while there are some basic guidelines, only your ear will tell you which mic position is best for the task at hand.

Of course, there are some basic tenets of the trade. Watch your gain staging to avoid noise. Avoid excessive use of EQ. Try to achieve a good sound during the tracking process itself. Make an effort to learn your room – how do your favourite recordings sound in it? Do you need to attend to your acoustics? Maybe throw some shelves or blankets up behind you to absorb or deflect soundwaves. Little things like these can make a big difference.

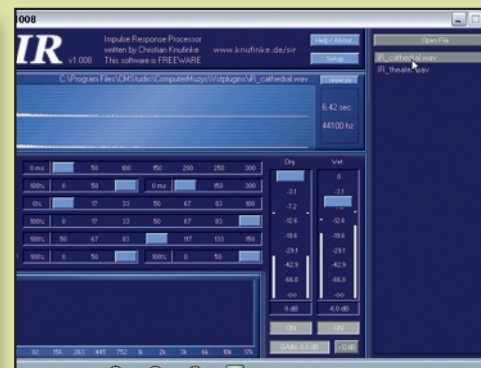
STEP BY STEP Using SIR



1 The freeware SIR impulse reverb is an excellent professional effects processor that offers levels of sound quality and realism that were undreamt of only a few years ago. Here we have a simple drum loop that was created using our own SR-202, and we'd like to add a sense of space to it. For this task, we'll call upon SIR, but you could use any other convolution reverb instead. >>



2 You can open an instance of SIR on either a channel insert or on a send. Normally, we'd recommend using an auxiliary send, since convolution reverbs can be somewhat resource-hungry. For demonstration purposes, we'll open it as a channel effect. You must also be aware of your host's ability to compensate for any delay added to the signal, as SIR can add a large amount! >>



3 Finally, use the plug-in's **Open File** function to browse for impulses. Click on it to hear the effect. Remember, a short delay will be introduced, so it will be a moment before you hear the results. Though there are only two impulses shown here, there are loads of free impulses to be found online. We've found everything from classic effects devices to wide-open and natural spaces.



Playback time

Q What is meant by plug-in delay compensation? I have a mate who just switched from Logic to Cubase SX because he said Logic didn't have it, and it was crucial to him. Do I need to worry about such things? I'm using Sonar, by the way.

Michael Wright

A When you record a signal into your computer, a slight delay is introduced (usually only a few milliseconds) while the signal is converted to digits and sent to and from your drive. This delay is called latency. The better your soundcard is, the lower this latency will be. Each real-time process that's applied to that signal also introduces latency. This includes plug-ins, such as reverbs, delays and the like. If you use a DSP card, such as the TC Electronic PowerCore or Universal Audio UAD-1, even more delay will be added as the signal is shuttled to and from the card. Some sequencers make allowances for these added latencies by calculating the added delays and adjusting the playback timing of all tracks accordingly. This process is referred to as 'delay compensation'. If your sequencer doesn't have this function, you have to adjust the playback timing of your tracks yourself. This can require a lot of complicated calculations, and is a real pain in the posterior. As your friend has noted, Logic doesn't currently have full delay compensation. However, it will still

▲ If you've got a PowerCore, your host had better have plug-in delay compensation or you'll wear out your calculator!

compensate for any delay added to an audio track insert – that, at least, is something.

Getting vocal

Q I want to begin adding vocals to my songs. I have an M-Audio Audiophile 2496 which has only the sort of inputs that you find on a home stereo. I borrowed a microphone from a friend, but it has those big three-pronged jacks. What do I need to do to record my voice? Do I need a mixer, or can I just buy an adapter?

Anthony Blackwood

A The inputs on your soundcard are intended for line level signals, which require no special amplification. While line level inputs are fine for synthesizers, effects devices and such, they don't provide the sort of gain that's needed for a microphone. For that, you'll need a preamp. Some soundcards have preamps built in, as do many mixers. Mackie mixers, for instance, are highly regarded for their onboard pres. A great pre can cost upwards of £1000, but there are serviceable units from companies such as PreSonus and ART for a lot less dough. You won't get the same sort of audio quality you'll hear on a world-class studio production, but you can still get the job done reasonably enough. If you're using a

condenser mic, it's also likely that you'll need to provide phantom power (more on that in a bit). Phantom power is sent from the mixer or preamp down one of those pins in the mic cable (XLR is the technical name for the type of jack you describe).

White noise

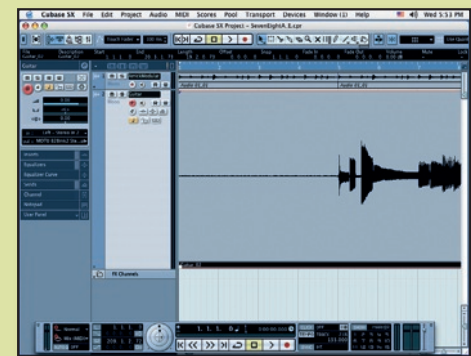
Q I've been buying the magazine for ages, and have learned a lot. However, I still find that my recordings are lacking. Most importantly, there's an alarming amount of noise in the signal. How can I avoid this, and what can I do about the noise on my existing tracks? I can't re-record some of this stuff.

James Brunt

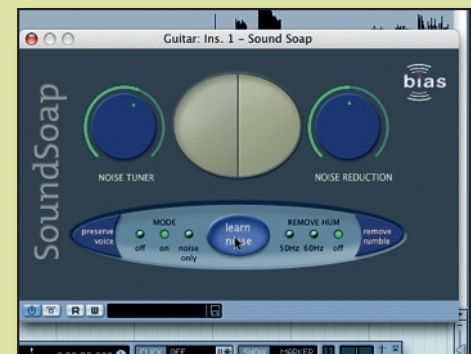
A To prevent excess noise levels in the first place, you must pay careful attention to gain staging. Every bit of gear you have adds a certain amount of noise. In fact, every single jack and cable in the chain will add noise. It's therefore crucial that you get the hottest level of signal into your computer without letting in too much excess muck. If the sliders on your mixer are pushed all the way up, but the gain level going into the mixer is low, you're probably adding lots of noise. Make sure the level going in is as hot as possible (without distortion) before you touch the faders.

When you have a noisy track that can't be re-recorded, try using a noise reduction application. These can work miracles, but should be used with discretion, as they can compromise your signal's integrity. Many such programs can analyse a sample of the noise by itself, then strip it from the program material. This can be a lifesaver when fixing up old tape transfers.

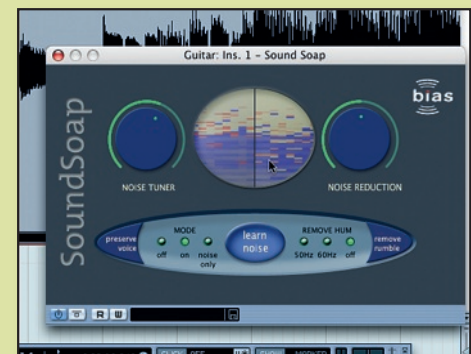
STEP BY STEP Reducing noise with SoundSoap



1 We're using Cubase SX here. We have a guitar track that has an unacceptable level of noise: this may be due to improper gain staging or inferior equipment. We could use EQ to filter out the noise, but that would compromise the signal. Instead, we'll call up BIAS's SoundSoap plug-in on one of the track inserts. >>



2 SoundSoap is capable of examining a sample of the offending noise and automatically removing it from the rest of the signal. This is done by playing back a segment of the noise and engaging SoundSoap's Learn Noise function. Fortunately, we've retained some of the noise that was recorded before the performance. >>



3 Once SoundSoap has learnt the noise, it can strip the offending frequencies from the material. This process is surprisingly effective: the recorded instruments come away intact, but sound as if the noise never existed in the first place.



▲ The PreSonus Firepod has quality preamps and sounds great

Limited use

Q What exactly is limiting? Is it like normalising, and when should I use it?
Chris Phelps

A A limiter is a type of dynamics processor: it's used to keep the signal below a given amplitude. In practical terms, a limiter is often used to prevent distortion or clipping on an input signal (such as from a microphone). The engineer sets a 'threshold' over which a reduction in level will occur. These days, limiting is often applied to an entire mix to make it seem louder, but this can easily be overdone, sometimes resulting in a lifeless track that is simply blisteringly loud.

Normalising, on the other

Pro Fools

Q I hear that an awful lot of producers swear by Pro Tools, but I can't understand why this system is better than simply using a computer and a soundcard. What's the big deal? I'm about to upgrade my system, and I'm wondering if Pro Tools is something I should be considering. Is its status just based on marketing hype, or am I missing something?
Kelly Mitchell

A Once upon a time, Pro Tools was the only system that allowed you to record and play back a significant number of digital audio tracks to and from a computer. Pro Tools is built on a combination of tried-and-tested

more and no less, and the system is very, very stable. This sort of dependability made Pro Tools the industry standard for professional recording and editing.

However, modern computers have increased in power at a staggering rate. Today's humble 'domestic' PC or Mac can offer dozens, even hundreds of simultaneous tracks of audio playback, and an incredible number of plug-ins, all running natively. With modern, stable operating systems like Mac OS X and Windows XP, we have far less hassle and downtime. They're not perfect, but even some pro studios are now banking on native systems such as Nuendo and SAW Studio.

Until recently, one reason to go with Pro Tools was for the sake of compatibility with professional studios. Yet, with affordable native versions of Pro Tools LE now out there, not to mention the now standard OMF file format, it's easy to take your music wherever you need to go these days.

Make no mistake: Pro Tools has a lot going for it. It's dead easy to use; there are some lovely plug-ins that are only available in one of Pro Tools' proprietary formats; and the power and stability of the system are truly drool-inducing. However, it won't, in itself, make your productions any more or less 'pro'.

Air head

Q It seems to me that I'd get better guitar recordings if I plugged my instrument directly into my soundcard rather than through my combo amp. Yet I have a friend who did some recording in a pro studio, and they always recorded his amp. Why is this better? I tried it, but I get a lot of noise.
Frank Moon

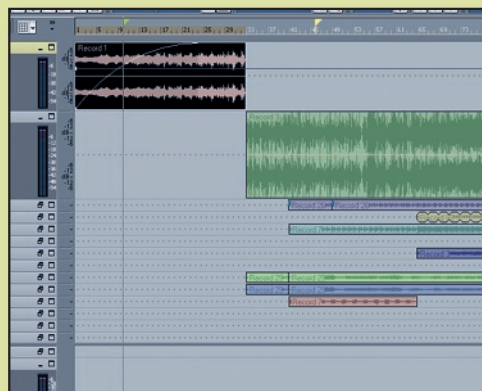
A 'Move some air' has become a kind of mantra in engineering circles. In our modern, software-based studios, it's possible for all of the sounds in a song to be generated from within your computer, then mixed and mastered natively. This means that your music never hits an acoustic environment until it's played back by the listener. While this approach may minimise unwanted noise, many feel you're missing something if you never get any air into your tracks. After all, sound is all about moving air. By pushing the air with an amp and then miking that up, you're adding real space to the sound. If you're sceptical, try splitting the signal and recording the direct and miked signals. You can choose how much of either to use during the mixing process.

A LIMITER IS USED TO PREVENT DISTORTION OR CLIPPING

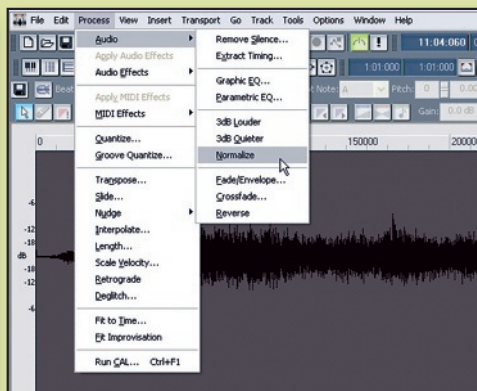
hand, is something that is used to raise or lower the level of the entire signal. This does nothing to the relationship between the quiet and loud bits, but instead brings the whole thing up (or down) so that the peak level is at a predetermined amplitude.

hardware and software. The hardware handles all of the DSP, and the software acts as a front-end. Since the hardware takes care of the processing, very little strain is put on the host computer. The user is guaranteed a certain number of tracks and plug-ins; they get no

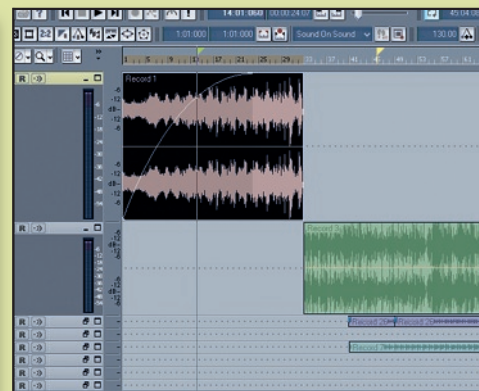
STEP BY STEP Normalising a track



1 Here we have a project open in Sonar 4. The first track is a bit too low in overall volume compared to the rest of the material. However, we'd prefer not to sacrifice the dynamics of the recording by applying compression or limiting. In a case like this, the solution is to normalise the audio file. >>



2 We can get a better idea of what we're doing if we use Sonar's built-in sample editor and loop construction view. Next, we need to go to the **Process** menu and select **Audio** > **Normalize** from the available options. Some audio editors will allow you to adjust the maximum dB for normalisation. >>



3 With the process complete, it's clear that the overall level of the entire audio file has been increased by a few dB, but the relationship between the loud bits and the quiet bits remains unchanged. Do watch yourself, though – if you apply any gain to a normalised file that's mixed full-volume, you'll clip the output.



▲ If your tracks are out of phase, you could wind up with reduced frequency content, or even silence. Check the phase in Nuendo with this extremely handy switch

A little phased

Q Could you please explain what phase means? When I click on the Phase button in Sonar, I hear no difference at all. What's the big deal here?

Rik Bergman

A Every sound wave has a positive and negative cycle or phase. When two identical signals are played simultaneously, but 180 degrees out of phase, they will cancel each other out, leaving only silence. Really! This means that an engineer has to be careful not to introduce phase-cancellation into the mix. For instance, if the engineer has miked both the front and back of the kick drum, the signals may be out of phase (since both will arrive at the recorder at different times). Even if

drum has vanished altogether when your track is being played out for the first time! So, check your phase during recording and mixing by clicking on that button or listening to a mono mix. Trust us, you'll be glad you did.

A new power

Q I just bought a used Behringer mixer with some XLR inputs. It also has phantom power, but I'm damned if I can figure out what that's for. What does phantom power do? Do I need it?

Marcus L

A Unlike a dynamic (or electret) microphone that retains an electrical charge, a condenser microphone needs to be provided with a power source.

CHECK YOUR PHASE DURING RECORDING AND MIXING - YOU'LL BE GLAD YOU DID

the signals aren't identical, they could still cancel out certain frequencies. You may not even know that there's a problem if you don't check the phase or audition your mix in mono. Many clubs play their tunes on mono systems... you don't want to hear that your kick

Typically 48 volts, this charge is sent down one of the conductors in the XLR microphone cable from the mixer or preamp. If you don't own a condenser microphone, it's unlikely that you'll ever need to activate the phantom power on your mixer. In fact, sending

phantom power down an XLR cable to a device that doesn't need it might damage it. If you aren't using an XLR cable, it's not really necessary for you to concern yourself with it.

Mic mayhem

Q I'm looking to buy a microphone for recording my voice and guitar. I see a lot of adverts for 'pro' mics that don't cost the earth - there seems to be an endless assortment to choose from. I am, to be honest, baffled. I don't understand the differences between dynamic and condenser mics. What sort do I need to look for, and are these budget things really professional quality?

Marty Klavuhn

A Dynamic microphones are the sort of handheld mic you might see on stage at a concert. These microphones are typically cardioid-only (meaning that they are engineered to pick up the signals directly in front of them while rejecting the sounds from either side and behind them), and may exhibit a frequency response that's tailored for on-stage use. (The upper-mids might be boosted to cut through a live mix, for example.) Some dynamic microphones are designed for a specific task, such as tracking kick drums or other instruments. A good, multi-purpose dynamic - such as the industry standard Shure SM-57 - is a good first microphone, as it'll still be useful to you when you've added more costly mics.

A large diaphragm condenser mic is ideal for vocals, as well as a great many other sources. If you can afford only one decent mic, consider a nice condenser. These mics are pretty sensitive and accurate, and a good one can make the difference between a track that sounds professional and one that sounds as if it was recorded in a closet.

A small diaphragm mic is usually geared to a specific purpose - recording acoustic strings, for example. This wouldn't

be an ideal choice if you're only buying one mic, as such hardware has a tendency to accentuate the high end a little.

The old adage about getting what you pay for applies perfectly to audio recording - perhaps more than to any other vocation. A great microphone can set you back a small fortune. However, the home recording revolution has narrowed the price-to-performance ratio across the board, and mic prices have tumbled. Even high-end companies such as Neumann and AKG have released budget-minded microphones: these sacrifice some of the more esoteric functions but cost significantly less than their high-end siblings. In addition, companies like Audio Technica and Rode have introduced a new class of low-cost microphone that has quickly become favoured by pros and amateurs alike. Audio Technica's 4033 has become an industry standard in less than a decade, but will only set you back a few hundred quid.

There are also plenty of very cheaply made, low-priced mics on the shelves. Many of these are assembled from generic capsules made in China, and may be of dubious build quality. Still, you might find they'll get you by if you're really strapped for cash. **sm**

► **Shure thing:** the SM-57 is an industry staple that's worth splashing a little cash on for any producer



COMING SOON: WINDOWS XP!

In issue 86, our crack team of computer music trouble-shooters will be solving your **Windows XP** problems. Mail your music-related Windows questions, no matter how basic or advanced, to ronan.macdonald@futurenet.co.uk.