



The World's Best Music Recording Magazine

Studio SOS (Through October 2005)

Article

Studio SOS

People : Studio SOS

October 2005

Phil Evans

This issue's Studio SOS comes to you from the other side of the world, as we help an Australian reader sort out his vocal recordings.

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September 2005

Jim Fish

The SOS team purge the unwanted buzzes and hums from reader Jim Fish's new studio.

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August 2005

Brett Taylor-Homes

The intrepid SOS crew answer a call for help from a drummer whose studio requires a monitoring and acoustics makeover...

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Jazz and Alessia

The SOS team return to Cambridge, turning another home studio upside down in search of improved vocal and guitar sounds.

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Rod Brakes

The owner of an unusually bijou studio setup provides the chocolate biscuits this month, as the SOS team get busy helping to improve the performance of his gear and the sound of his recordings.

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East Norfolk Sixth Form College

East Norfolk Sixth Form College needed help integrating their hardware multitracker with their computer sequencing system, so the SOS team travelled over to Great Yarmouth to lend a hand.

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Above & Beyond/Anjunbeats

The SOS team are back in London this month to help some high-profile remixers sort out the monitoring problems in their new studio.

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Tom Lindsey

The SOS crew travel to an attic in Birmingham (UK) to help a drummer improve his recordings.

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Aniff Akinola

Acoustic problems and mains hum were making Aniff's studio difficult to use, so the SOS team stepped in to help.

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The Loose Cannons

The SOS crew head to the 14th floor of a London tower block to help a pair of readers improve their drum sound.

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Rob O'Neil

A dodgy snare recording was just one of the problems facing Rob O'Neil at the mixdown stage, so the SOS dynamic duo (Paul and Hugh) visited his home studio to help him sort things out.

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November 2004

Tony Global

Tony Global's background is in the dance-music/DJ scene. The SOS team help him to polish his vocal sounds and improve his mixes.

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Dorian Kelly

The SOS team ride to the rescue of a budding media composer who is having trouble with his mixes.

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TV Composer David Lowe

This month, we help television composer Dave Lowe transform a cavernous-sounding spare bedroom into a usable home studio.

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Pete Keen and Nick Smith

Setting up your studio in a cube-shaped room isn't a very good idea, as Pete Keen and Nick Smith found out to their cost. So the SOS team set off to Kidderminster to help find some solutions to the inevitable acoustic problems.

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Peter May

The SOS team help Peter May to brush up his drum sounds and put more life into his mixes.

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The Arcades

This month, the SOS team help The Arcades to rock even harder than before!

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Jiang Li

The SOS team apply themselves to the task of mixing Chinese traditional instruments with mainstream Western sounds at reader Jiang Li's home studio.

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Chris Brockis

Chris Brockis had been having trouble mixing in his attic home studio, so he enlisted the help of the trusty SOS team.

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Nigel Helm-Nurney & Pombokiwi

The intrepid SOS team travel to Nigel Helm-Nurney's London studio to help his band Pombokiwi do battle with unwanted spill, uninspiring guitar sounds, and masses of egg boxes!

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David Ashman

David Ashman felt that his mixes lacked energy and were sounding 'too digital', so the SOS team set off to his home in Bristol to sort out his monitoring system and mix processing.

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Glenn Bucci

During the October AES Convention in New York, the SOS team managed to make it to Amityville to help reader (and SOS Forum regular) Glenn Bucci with his recorded guitar and bass sounds.

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Clive John

Recording piano was turning out to be a grand challenge in Clive John's home studio, so the SOS team headed over to sort things out.

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The Harding Family

Musicality isn't everything when you're trying to make a decent-sounding recording at home, even if you're as talented as the multi-instrumentalist Harding family. So the SOS team travelled north to help them get their engineering techniques up to scratch.

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Gordon Giltrap

The SOS team visit the West Midlands, where Gordon Giltrap's home studio needs help, nestling as it does in the shadow of one of the UK's most powerful TV and radio transmitters.

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September 2003

Alistair Vickery and Jon Midwinter

The SOS rescue team set off for Bristol to help out readers Alistair Vickery and Jon Midwinter, who wanted help with their mixes — not only getting the bass sounding right, but also stopping it getting out of the house and annoying the neighbours!

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July 2003

Allan Murrell

This month the intrepid SOS team travel to Wigan to address Allan Murrell's recording, monitoring and mixing problems.

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The SOS team battles through snowdrifts to help Nick Redman and Mike Sinnott with two different vocal sounds.

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Dave Stevens' home studio

This month the SOS team visit reader Dave Stevens' home studio to sort out a chesty vocal sound, investigate a mystery digital buzz, and hand out some mastering tips.

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Tom Fox

Recording in a converted attic, Tom Fox was having serious problems with his acoustics while recording drums, so the SOS team drove over to Yorkshire to sort things out.

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Alan Pittaway

This month SOS helps Alan Pittaway improve his drum kit recordings, and also helps him get his two Roland multitrackers working together.

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Noor Ali

Hums, buzzes and noise were stopping Noor Ali recording his guitar parts, so SOS headed over to Worcestershire to set things straight.

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November 2002

Murat Yucel

A special holiday edition of our hands-on troubleshooting column comes direct from sunny Turkey, where Paul White forsakes the beach to help Murat Yucel refine his band's recording setup.

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Nick Tucker

The SOS team rushes to the rescue of a reader in Somerset suffering from boxy vocals, a weedy mix, and a dodgy tweeter.

STUDIO SOS

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September 2002

Tim Way

Another reader's studio gets the benefit of expert SOS staff attention. This month, it's the turn of Tim Way, whose mixes sound fine in his own studio, but don't travel well.

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August 2002

Steve Graham

The SOS team answer a reader's cry for help, and assist him to improve his recordings of guitar and vocals.

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The World's Best Music Recording Magazine

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Studio SOS

Phil Evans

Published in SOS October 2005

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This month's Studio SOS comes to you from the other side of the world, as we help a reader in Sydney sort out his vocal recordings.

Paul White

If you were to drill straight down into the floor of the SOS offices in Cambridge, around 7000 miles later you'd emerge within spitting distance of Phil Evans' private studio, Chorus Music & Media, but in the interests of health and safety, we opted to fly. Phil is a full-time professional musician who runs his studio from his home in Wahroonga, a northern suburb of Sydney, working on his own material as well as demos for friends and clients. Phil was a *Logic* user until Apple discontinued the Windows version of *Logic*, since when he's switched to Steinberg's *Cubase SX 3* and is finding some aspects of the transition quite difficult to come to terms with. He prefers *Logic*, but because he relies on other Windows applications, he's not prepared to move over to a Mac.

His only keyboard is a Korg 01W workstation and he's recently purchased an Echo Layla 3G audio interface to complete the system. His PC has had some cooling modifications to make it quieter, though not as quiet as he would like, and his Alesis Monitor One MkII speakers are driven by a beefy Sansui AU317 hi-fi amplifier. Currently, Phil is working with singer Jessica McPherson, and was keen to see if he could improve on the vocal sound he was getting. He'd also experienced a few little technical problems that he needed help with.

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Monitoring System Tweaks

While Phil was working on making the first cup of tea of the day, I checked out the monitoring system. The Alesis monitors were set up on a shelf above the control desk facing straight ahead, and they were also a little too high for Phil's seated listening position, so I suggested we angle them in and down so that the

tweeters faced directly towards his listening position. We achieved this using some small blocks of wood to prop up the rear of each speaker, and to my ears this improved the imaging, giving a better phantom centre image and a wider sweet spot.

The room itself was around five metres square and about half that in height, which is less than ideal for a studio, as all the room modes tend to pile up at around the same frequency causing large peaks and dips at low frequencies. Phil had tried to improve this by screening off about one metre of the back of the room using heavy curtains, with sheets of foam and other material suspended behind. As the Monitor One MkIIs don't have huge bass extension, the bass end turned out to be reasonably even at the listening position, though there was the inevitable dead spot close to the centre of the room, where the bass dropped in level. A large foam upholstered corner sofa also helped soak up unwanted audio energy, and as the side walls were quite a long way from the speakers, it didn't matter too much that these were reflective rather than absorbent. A square metre or so of acoustic foam on the side walls opposite the listening position might make a small improvement, but the room turned out to be quite workable. I did however suggest that if he upgraded to larger monitors with a greater bass extension, then he should consider stacking a few rolls of rockwool loft insulation behind the curtain at the rear of the room to help with the bass trapping.



Getting a good vocal sound required draping a duvet behind the singer and taping a pillow to the mic stand — this reduced the level of room sound reaching the mic, resulting in a less coloured and more usable sound.

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Hunting Down Audio Clicks

Phil's main technical problem was that he'd noticed occasional clicks when trying to record the audio outputs of his Korg synth, set to a piano sound at the time, but in fact the clicks affected all the synth sounds. His sequencer's audio interface was set to use a buffer size of 256, and the audio levels were nowhere near clipping, so I suggested he try the variable-gain mic/line inputs on his Echo interface just to check whether there might be something odd about using that particular synth with the Echo's line ins. As I half expected, this turned out to have no effect on the clicking problems.

Playing soft synths via MIDI showed up no such problems, so it seemed that only the audio input was being affected. We also tried feeding the Korg via a small Behringer mixer before going into the Echo interface in case there was a matching issue, but that didn't help either. We checked the headphone outputs on the Korg and on the Behringer mixer and there was no problem there, so that ruled out a weird clipping problem that the meters weren't showing up. I got Phil

to switch off his graphics card's accelerator mode, as that can sometimes cause problems, but again to no avail.

Phil was at first pretty sure that the clicks didn't appear when recording vocals, but when we ran some tests to confirm this, we also found clicks, and examining the audio waveform in *Cubase*, we discovered narrow spikes. Subsequent enquiries to Echo revealed that the driver exhibits a bug that causes clicks when the unit's MIDI interface is also being used, but they told Phil that they expected to be releasing a fix within a week or so.



Phil's compression setting, shown here, was already very suitable for adding body to Jessica's well-controlled vocal delivery.

Another mystery was a high level of digital noise and hum audible over the monitors, but only when *Cubase* was running. Other audio applications, such as *Mackie Tracktion* and *Adobe Audition* seemed to work without causing noise. This noise didn't get onto the mixes, which Phil did via internal bounces, but it was distracting when mixing. It will be interesting to see if this goes away when the new Echo driver becomes available.

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Recording Vocals

The next stage was to move on to recording some vocals, with Jessica sportingly obliging to be our guinea pig. Phil had been recording with the mic facing his curtain, which cut down on some unwanted reflections, but there were still noticeable reflections from the side wall. He'd also set up a lightweight oriental-style folding screen behind the mic to try to isolate the computer noise, but this was really too light to have any significant effect.

To try to improve the situation, I draped a large duvet over the screen and placed this against the side wall to create an absorbent corner. The curtain room divider formed the other side of the corner. Jessica sang with her back to the corner, but to further reduce reflections and sound from elsewhere in the room I also taped a pillow to the mic stand, behind the mic — a Rode NT1000 — which I turned upside down so the pillow's acoustic shadow would fall in the right place. It might not have looked very sophisticated, but we got a



great dry vocal sound straight off. Jessica has a fairly bright voice, and the NT1000, having a significant presence peak, produced a slightly aggressive sound, though nothing too serious — we could have got a usable sound with a bit of careful EQ or mic repositioning if we'd had to. However, we also had a Rode NT2A available, which is actually a very different microphone, with a smoother overall sound, and with Jessica's voice this produced a sweeter, more musical sound that we could use with very little additional EQ.

The compression setting that Phil was using was already fairly good, with a 4:1 ratio, a 10ms attack time, and a 100ms release time. I readjusted the threshold setting to give 4-6dB of gain reduction and it sounded fine, adding density without otherwise compromising Jessica's excellent and well-controlled voice. A little EQ, pulling back at around 2kHz, dealt with a little remaining mid-range harshness and I added just a hint (1.7dB) at 10kHz to create the contemporary 'air' effect. EQ above 10kHz is usually well above the area where singers tend to have their natural presence peak, so it tends not to make the sound any more aggressive. At this point we had a great sound that only needed reverb to make it sit properly in the mix.

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Phil's Comments

"Thanks so much for your visit to my studio — I'm still buzzing! I really did find a lot of your tips very helpful, and after you left Jess and I recorded a short demo of some National Anthems for her to audition for a rugby gig, and I really was pleased with the vocal sound. I noticed several more spikes in the track, and must confess that I have recorded many other tracks with the same problem but have had to work around it by recording small sections at a time and editing them together. Let's hope the new driver sorts things out!"

 www.chorusic.com.au

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Reverb Treatment

Phil had downloaded the new Steinberg *Roomworks* reverb plug-in (from www.steinberg.net) and I was keen to try this as I hadn't had the opportunity to use it before now. Phil had tried a number of traditional reverb sounds, but as so often happens they had pushed the vocal back in the mix and tended to fill up space in the mix. I tried a few of the presets and eventually settled on High Frequency Reverb as a starting point, setting the delay time to 2.12 seconds with 60ms of pre-delay. I dropped the diffusion to




Once the compression and EQ had been sorted out, Paul set about creating a suitable vocal reverb. For the more uptempo songs, a shorter 'early reflections' style patch (above) was used, but this was chained with a

11 percent, set the high-frequency damping to 126 percent and the low-frequency damping to 10 percent, the

brighter and longer patch (below) for the ballads.

aim being to get a more ambient, 'early reflections' type of effect. With a reverb mix of around 20 percent, this produced a very nice, up-front sound with a good sense of space but with no clutter — almost like a bright ambience program. Everyone agreed the result sounded much more contemporary and 'right' than the previous more obvious reverb setting. I thought this new *Roomworks* reverb plug-in was noticeably better than the old one, and as it's apparently a free download for *Cubase* users, anyone else dissatisfied with their existing *Cubase* reverb might want to give it a try.

On a previous Studio SOS visit, I combined two *Cubase* reverbs to create an effect that I couldn't get from one on its own and decided to try that here to see if we could conjure up a reverb suitable for some of Jessica's more ballad-like songs. We kept the reverb I'd just come up with but followed this with a bright plate patch using a 3.22-second delay time and a pre-delay of 115ms to keep it out of the way of the early reverb. With a balance of around 25 percent, this added a low-level reverb tail without clouding the sound, and produced a nicely controlled sizzle at the ends of words and phrases. Phil liked both these settings and saved them. On balance, Phil was pleased with the low-tech improvements we'd made to his vocal recordings and he has since made other recordings using this system and reported good results. 

Published in SOS October 2005

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Studio SOS

Jim Fish

Published in SOS September 2005

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The SOS team purge the unwanted buzzes and hums from reader Jim Fish's new studio.

Hugh Robjohns

This month's Studio SOS visit came about because of a post on the SOS Web Forum in which reader Jim Fish complained of serious hum problems in his studio. He had recently moved house and was in the process of rebuilding his studio equipment into a large room above a timber-framed garage, but he had started to get such high levels of noise that it could clearly be heard in the next room! He was suspicious that the problem might have been caused by two new LCD computer screens, with which he had recently replaced the previous pair of 'legacy' CRT monitors.

He described his equipment as being based on a Dell PC coupled via Firewire to an original MOTU 828. Mixing was performed on a large Topaz eight-buss analogue mixer and monitored on a pair of Fostex PM2 active nearfields, with a range of outboard effects units and synths hooked into the system via various patchbays. His pair of 18-inch LCD screens were mounted on the wall behind the Topaz desk and hooked directly to a Radeon dual-head graphics card in the PC.

The symptoms of the problem were intriguing, as the hum apparently remained if



Jim's studio system was picking up unwanted noises from his new flat-screen monitors, but Hugh wasn't sure whether this was a problem with the LCD screens themselves or whether it indicated an underlying earthing problem. To test this, he powered a separate laptop, Jim's MOTU 828 audio interface, and one of the Fostex speakers from a separate plug socket. This test system picked up none of the noises which the main system was suffering from, indicating that the problem stemmed from earthing problems.

the mixer or MOTU 828 were turned off, and it became louder if the Firewire link between computer and monitors was removed. The only thing that silenced the hum was turning off the LCD monitors or removing their video inputs (in which case they shut down anyway, of course). One monitor seemed to create a louder hum than the other, and they produced slightly different pitches too!

Jim had found that jiggling the monitor video connectors or power leads affected the hum (suggesting poor earthing perhaps), and after relocating the monitor VGA cables along the back wall (to move them away from mains and audio cables) the hum level was reduced — although that could just have been a coincidence of remaking the connections, and thus improving the earthing contacts in the process.

A number of other Forum posters had offered help, but to no real avail. The usual suggestions of moving the LCD screens' power supplies around, using balanced connections to the Fostex speakers, and changing the computer graphics card had already been tried and found to make no difference. Questions over noisy mains supplies appeared to have already been addressed as well, since Jim was feeding all his equipment via ETA power conditioners. Clearly, this was a complicated problem and it sounded like a suitable challenge for Studio SOS.

However, when Reviews Editor Mike Senior suggested a visit, Jim confessed to being a 'Hobnob scrooge', and we all know that the Paul White doesn't get out of bed nowadays unless there's a pack of chocolate Hobnobs in it! Fortunately for Jim, though, it turned out that Paul was going to be away anyway visiting a certain Antipodean microphone manufacturer, and since Mike doesn't suffer the same debilitating condition as our Editor In Chief, it fell upon him to join me in attempting to find a solution to this thorny problem.

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SOS To The Rescue!

On our arrival in Bungay, Jim demonstrated the loud buzzy hum he was getting on the system, and how it disappeared completely if he turned the LCD monitors off. That suggested that the buzz was definitely being generated by his two LCD screens, but exactly where the noise was getting into the system was far from obvious.



To confirm the source, we substituted the two GNR-brand screens for a different LCD screen of a different make, borrowed from his office, and that produced a substantially lower level of buzz and noise. However, the

replacement monitor wasn't capable of the very high screen resolutions that Jim was using on his two GNR monitors, and it is certainly possible that the additional demands placed on the monitor and its power supplies to provide such high screen resolutions were a significant factor in the level and quality of noise generated. Jim was very keen to maintain the current high screen resolutions, though, and, while switching to less demanding resolutions may have helped, it clearly wasn't a proper solution to the problem.



The first problem in Jim's earthing system was quickly spotted when the SOS team began poking around under the main studio desk — one of the mains distribution boards had a European mains plug which had been forced into a UK socket. This effectively lifted the earth connection to the audio interface and mixer, not only rendering the studio potentially lethal, but also contributing to the noise problem. Mike fitted a new UK plug without delay!

Although it had already been tried, our first step was to experiment with moving the LCD screens' power supplies around. The specific angle, orientation, and distance of these separate switched-mode power lumps from other equipment and cabling can make a huge difference, as they usually emit electromagnetic radiation in specific patterns. However, after carefully turning them through 360 degrees in all directions we found no significant difference to the level of buzz. So direct radiation didn't appear to be the cause, and it looked much more like a complicated ground-loop problem.

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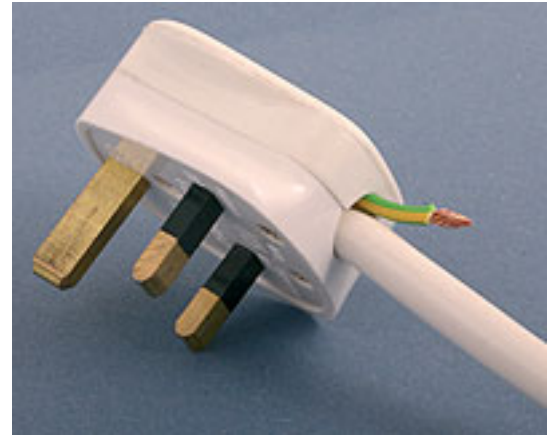
Tracking Down Earthing Problems

The key to solving any kind of ground-loop problem like this is to work through the system logically and patiently. So the first step was to unplug the audio input to the Fostex speakers to make sure that they weren't picking the noise up directly through their mains power supplies, or via radio-frequency interference. Sure enough, the hum disappeared as soon as both speaker inputs were unplugged, proving that they were not faulty and that the hum was reaching them as a genuine audio signal.

At this stage, given the nature of the buzz and the confusing symptoms, it seemed fairly certain that the problem was the result of one or more earth loops. Jim had a lot of equipment hooked up to the desk, some balanced and some unbalanced, and the rat's nest of mains, audio, and computer cabling under the large custom desk probably wasn't helping either. So we decided it would be worth proving whether or not the suspect LCD monitors could be used in a minimalist system without creating the excessive hum apparent in the full system.

We disconnected the MOTU 828, one of the Fostex speakers and one of the LCD monitors from the main system, and re-connected them to a laptop

computer, all powered from a separate mains extension fed from a different wall socket. An audio connection between one of the MOTU outputs and the Fostex input created no buzz at all — so we could now be confident that, while the LCD monitors were undoubtedly the source of the buzzing noises in the main system, sensible mains earthing and lack of ground loops could produce a quiet system.



Another lifted mains earth connection was discovered when Hugh spotted an earth wire protruding from the casing of a plug — a common way of indicating that an earth wire has been disconnected. Again, this earth wire was re-connected to avoid the risk of electric shock when using the studio.

Next we re-plugged the speaker, interface, and screen back into their original mains sockets under the desk... and the buzz returned! We hadn't changed any signal connections, so we hadn't created a ground loop that way, and the implication was therefore of a problem in the mains power distribution for the main system. A quick glance under the desk revealed an evil tangle of mains and audio cables, and Jim also admitted that the equipment was fed from two different wall sockets on opposite sides of the room. At this point we politely suggested a bit of rewiring...

Jim had no objections, so we were soon scrabbling around under the desk — a regular feature of Studio SOS visits! After a bit of pruning back of audio cables to reveal the mains distribution boards we spotted the first problem. A European mains plug had been forced into the first four-way plugboard coming off the left-hand wall socket. It turned out that this was feeding a (metal cased) switched mains distribution unit bolted into the left-hand desk rack — a unit purchased from a European supplier. This distribution panel was feeding both the MOTU interface and the Topaz mixer, amongst other things, and although the European plug was equipped with an earth terminal, the action of plugging it into a standard UK mains distribution board meant that it was left dangerously unearthed. So, neither the MOTU nor the console were actually earthed properly, which not only made them potentially lethal, but also made them a significant contributing factor in the buzz problem.

Mike volunteered to replace the European mains plug with a standard UK-style plug, and five minutes later when we re-powered the system the level of buzz was immediately improved, although it was still unacceptably audible. We were making some progress, and the system was definitely safer now, but clearly there were still some further problems to resolve.

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Re-configuring The Mains Wiring

Since I was already under the desk, I set about unplugging all the equipment mains plugs and rewiring everything in a more sensible 'star' configuration, stemming from the wall socket on the left of the studio. All the ancillary equipment — lights, phone chargers, and so on, were fed from the right-hand wall socket to keep them from degrading the mains quality of the audio setup.

While doing this, we came upon a second bit of comedy mains wiring — the mains plug of a Roland U110 sound module had its earth wire protruding from the plug casing. Someone had obviously isolated the earth wire, probably in an uneducated attempt to cure a hum problem. At least they'd had the sense to leave the earth wire showing so that we could all see that it was in a dangerous condition! I re-connected it and made a mental note to check on the hum level when we got the rest of the system sorted out.



Once the mains cables had been re-organised, the audio wiring was tied up out of the way of them using cup hooks and Velcro cable ties.

At the same time as rewiring the mains, we also re-patched the audio cabling to the rackmounted equipment and suspended it all well away from the mains wiring using a cup hook and some velcro strips. Several eight-way snakes had been used to hook up the console inputs and outputs with the MOTU and patch bays, and in places these were coiled on top of mains cabling and power-supply transformers, which wouldn't have helped the system noise floor.

So, with the mains distribution re-organised, audio cables re-routed, and the MOTU, LCD screens, and Fostex speakers reinstated to their original configurations, we fired everything up again and had a listen. The bad news was that the buzz was still there, but the good news was that it was even quieter than it had been previously. The restructuring had obviously improved things further, but there was still a major problem somewhere.

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Hunting Down The Final Earth Loop

After further head-scratching, we returned to the idea that there must still be earth-loop problems, and our suspicion initially fell on the right-hand side of the studio, where a number of unbalanced rack synths were being fed via a balanced patchbay to the mixer. However, removing the eight-way multicore link between the patchbay and desk made no appreciable difference to the level of buzz.

However, coincidentally at this point we noticed that the level and quality of the buzz varied a little if the Firewire or mains connections at the back of the MOTU interface were wiggled. Furthermore, we found that the buzz increased

dramatically in volume if either the Firewire or mains cables were removed — something Jim had mentioned in his original forum post. Mike felt that this pointed the finger at possible earthing problems in the Dell PC, as it suggested to him that the computer was relying on the MOTU for its earth connection. We tried grounding the PC via a jack cable to the mixer (with the MOTU Firewire cable disconnected), but that yielded no improvement. We were also able to exonerate the PC by replacing it with the laptop — which also made no difference to the buzz.

With the computer re-connected, our attention finally turned to the potential for earth loops created via the MOTU's audio connections. The first step was to unplug all the audio cabling between the MOTU and the Topaz desk (we had already removed the synth rack connections), and that resulted in blissful silence. So the desk coupled to the Fostex speakers formed an inherently quiet system, which was encouraging.

Plugging the audio cabling from the synth racks on the right of the studio reinstated a very low level of buzz, and this was traced to one particular cable fed from one channel of an Emu sound module.

The other channel was quiet, and the noise varied in level as the plug in the back of the unit was wiggled, so we suspected that the output socket was making a poor earthing connection — perhaps one of the internal contacts was bent or slightly corroded or dirty. This was a relatively minor problem, though, and the buzz was barely audible once we'd fiddled with the connections a bit. In the longer term the Emu unit would benefit from having the socket cleaned and/or replaced, but it obviously wasn't the cause of the annoying buzz problem.

We continued re-plugging various cables back into the mixer until we reached the loom carrying the eight outputs from the MOTU interface, which plugged into the first eight mixer channels. Everything remained delightfully free from hum until the moment we inserted the first of these plugs, when the buzz was immediately reinstated in all its glory. It looked, then, like the problem was one of a ground loop with the MOTU connections, and I suggested the simple test of lifting the earth in one of the audio cables between the MOTU and the desk, to see if that would solve the problem, given that these connections were balanced.

Mike had brought a DI box with him just in case we wanted to try something like this, so we connected one of the MOTU's outputs to the DI box input and its XLR output back to the desk. Flipping the earth-lift switch on the DI box instantly cured the buzz, without affecting the signal quality at all, so we had finally found the



Although the re-organisation of the studio wiring had reduced the levels of unwanted noise significantly, there still appeared to be earth-loop problems causing hum on the system. The noise remained even when the computer and all the sound modules were completely removed from the system. The shadow of suspicion therefore fell on the mixer and audio-interface connections.

ground-loop culprit.

A permanent solution would involve making or modifying a cable loom with the screens disconnected at the console end. All of Jim's ready-made looms had moulded connectors, which were impossible to modify in the way we needed, but he was able to find six individual balanced cables with Neutrik jack plugs on the ends, which we could dismantle and modify fairly easily. I showed Jim how to snip the earth connection at the desk-input end of each one, and we re-plugged the MOTU using these modified cables. With all six connected the buzz was completely gone, and on playing back one of Jim's mixes from the computer everything emerged in a nice clean state — much to everyone's relief! Jim resolved to locate another couple of cables and modify them to isolate the screen at the desk end after we had gone, completing the setup.



The hum finally vanished once the final connection between the mixer and the audio interface was unplugged, which suggested that the problematic earth loop was being created between these two pieces of equipment. Re-connecting a single audio cable via a DI box to lift the signal earth confirmed that this was indeed the problem.

As Jim was still in the process of expanding his studio we had a few further suggestions for him to consider. Firstly, any other connections made to the mixer (he was planning to rig up the MOTU's inputs as well) which reinstated the hum should be tackled in a similar way. Secondly, if connecting any unbalanced gear to the mixer caused hum, then isolating transformers or DI boxes would be needed. In neither case should any mains earth be disconnected!

The synths on the right of the studio desk were being connected via a balanced patchbay to the balanced inputs of the desk, but many of the synths had unbalanced outputs. Sometimes this arrangement can create a 'one-legged' unbalanced signal for the desk, with low signal levels and obtrusive noise as a result. Proper unbalanced-to-balanced (pseudo-balanced) cables between the unbalanced synth and balanced patchbay would avoid this potential problem.

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A Job Well Done

All in all, this had been a very challenging problem, with a number of interacting issues combining to create a seemingly illogical situation. The lack of a proper mains earth on one of the mains distribution units — because of its European plug — was worrying and proved to be a significant facet of the buzz problem.

Other aspects included the unstructured mains distribution, the close proximity of audio and mains cabling, and the ground loops between the MOTU 828 audio interface and the mixing console. Clearly, the LCD screens were the source of the buzzing noise, and different monitors might have alleviated the problem completely, despite the other wiring issues, but with some logical experimentation and attention to detail we were able to cure the buzz completely and leave Jim with a clean and quiet system. **SOS**

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Jim's Comments

"After a long day of head-scratching the guys did a fantastic job and got the buzz so it was nestled right down with the hiss from some of the effects units and such (which is only audible when the system is turned up twice as much as it would ever need to be). Well, as you can imagine, I was over the moon with this result, after two months of crippling my ear drums and literally giving myself a migraine whenever I



tried to work on something. For the next week I was in a state of bliss, and I was finally able to get back to work. Unfortunately my happiness was short-lived, because the buzz crept back, although luckily still not even half as bad as it was before — I could just about keep on working. A few weeks after this, though, I got so peeved with my Dell PC for its constant crashing that I replaced it with a new custom-built model. Guess what? No buzz! Hooray! So the problem is finally solved, but really the whole thing still seems totally illogical. Encouraged only by my girlfriend, who likes to treat an illogical problem with an illogical solution, I have come to the conclusion that it was definitely the pixies playing tricks on me... yes, the pixies... that's what it was..."

Published in SOS September 2005

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SOUND ON SOUND

The World's Best Music Recording Magazine

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Studio SOS

Brett Taylor-Homes

Published in SOS August 2005

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People : Studio SOS

Sorting out the monitoring in a studio with carpeted walls is the challenge for the SOS team this month.

Paul White

Brett Taylor-Holmes called SOS because he'd just finished building a studio in a single-floor extension to his Herefordshire home and had discovered that the monitoring acoustics weren't all he'd hoped for. He'd also bought a Mackie Control to go with his Apple *Logic*/G5 setup and had problems getting it to work. We arrived at Brett's rather nice period house near Malvern, which adjoins the local cricket green, and were served tea and Hobnobs while we asked more about Brett and his studio, which he wants to use mainly for composing film and TV music, as well as for video editing. Brett is currently taking a master's degree in film making and is also a performing drummer and drum teacher.

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Too Much Carpet!

As soon as we entered the studio, we could see some of the potential trouble spots. The monitors, which were Mackie HR824s, were set directly on top of a large, flat desk and were positioned well back in the corners where they were bound to generate strong reflections from the surface of the desk. Perhaps more



Paul constructs improvised bass traps by cutting an Auralex foam panel in half and then mounting small foam blocks to the halves to distance them from the surfaces to which they are to be fixed.

seriously, Brett had used thin carpet to line the walls of his studio in the hope of providing a less reflective environment. In small areas carpet can be useful, but because it is so thin it only absorbs high frequencies with any degree of efficiency. Low and mid-range sound simply passes through it to the walls behind and then bounces back into the room, the result being a dry top end with a boxy mid-range.

He'd treated his live room in the same way with much the same result, and the door between the live room and control room was a lightweight interior door which didn't seal all the way around, so isolation between the two rooms wasn't great. This wasn't a problem when recording drums, as Brett uses his Roland V-Drum kit for that, but when recording vocals he'd had trouble with spill from the control-room monitors getting onto the recording. This was clearly going to take lots of Hobnobs!

Before looking at how to improve the acoustics, Hugh and I both felt the monitors should be brought much further forward to move them out of the room corners, bringing them in front of the computer monitor screens and helping to reduce the significance of desk reflections. We also positioned them on resilient pads to stop vibrations getting through into the desktop.

Listening to some pre-recorded Donald Fagen tracks from CD quickly confirmed that something needed to be done. Amongst the bits and pieces kindly provided by Auralex, we had a set of foam speaker pads which we used with their extra wedge pieces inserted to get the speakers roughly parallel to the desktop, as this gave the best tweeter height to match Brett's normal seating position. We brought the speakers forward to reduce the amount of desk area in front of them as well as to prevent diffraction from the LCD monitors. In doing so we had also moved them away from the corners, so we looked around the back of the HR824s to see what EQ switch settings had been used.

Brett had the monitors set up for 'quarter space' which was correct for working close to corners, but we switched them to 'half space' now that they were much further into the room. He'd also left the bass extension at maximum, which provided too much low end for such a small room — especially one with no real bass trapping. Switching to the middle 47Hz setting improved things, and listening to the same record produced a tighter, more even bass end, but with very muddled stereo imaging and still some residual lumpiness in the bass. We also noticed that the bass tended to disappear as you moved back from the listening position, something we've noticed in a lot of small rooms. Careful trapping can help, but in most cases you simply have to identify the rogue areas and keep away from them when mixing.

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Improving The Stereo Imaging

To sort out the worst of the imaging problems, we used three panels of acoustic foam, one on either side of the listener at the 'mirror' points, and one on the

ceiling. If you hold a mirror flat against the wall and mark where you can see the monitor from your normal listening position, that denotes where the centre of your acoustic foam panel needs to be. The trick works for placement of the ceiling panel too. Brett had placed a window through to the live room at the left-hand mirror point, and there was a large picture behind glass on the other, which was less than ideal! So after removing the picture, the right-wall and ceiling foam panels were fixed in place using spray adhesive, but the left-wall panel was temporarily fixed over the control room window using masking tape. Our suggestion for the long term was to glue this panel onto an MDF sheet and then hang it over the control-room window on hooks only when mixing. During tracking it could be left off to restore sight lines into the live room.

The back of the room was another flat carpeted wall, so we decided to use one of the new Auralex four-inch panels that comes with thick foam spacing blocks which allow it to be used away from the wall to increase the low-frequency absorption. We put this right at the top of the rear wall, since you need to get the trapping into corners to have any impact on low end. Although we didn't have any corner bass traps with us, we told Brett that it wouldn't hurt to add some later — there was plenty of space to install something in the two rear corners of the room, which would be ideal. The panel kit we had used comes with a pair of polystyrene diffusers which we felt might also help if we fixed them to the rear wall below the foam to scatter some high end, rather than allowing the carpet to absorb it all. The idea was tested subjectively by temporarily propping them in place and conducting more listening tests. There was a small but worthwhile increase in liveliness which further countered the tendency of the room to sound boxy, so we decided to fix them permanently using the spray adhesive again. A word of warning here — polystyrene reacts very badly to some types of glue, so check the glue on a gash piece of polystyrene first to avoid having to watch your new diffusers melt away before your eyes!



To get better response from Brett's Mackie HR824s, Paul & Hugh moved them away from the corners of the room, and placed them on some foam isolator pads. Acoustic foam was then placed on either side of monitoring position to reduce reflections which were muddling the stereo image — although on the left side the panel was only fixed temporarily so that it wouldn't obstruct the sight line into the live room while tracking.

We had one more four-inch foam panel left which we decided to cut in half and fix across the front corners of the room using the supplied spacing blocks, which happen to be cut at a 45-degree angle for exactly this purpose. We could have installed them directly behind the monitors or up against the ceiling, but we decided on the latter as it should make them more effective at low frequencies — and it also looked better.

While these little modifications weren't going to transform Brett's studio into a

world-class monitoring facility, the subjective improvement was very significant. Now the stereo imaging was good and the room sounded much more neutral and far less boxy. It also looked good, which is important if you're spending long hours working in one place.

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Live-room Acoustics

Turning our attention to the live room, it became clear that the dividing wall was a simple stud partition, with a single layer of MDF on either side and no insulation between them at all. So sound isolation between the control and live rooms was very poor, and only a complete rebuild of the wall would help — which wasn't practical or necessary. We felt that the best approach would be to make the best of the situation by modifying the way Brett worked when tracking, and to improve the live room acoustics to help with vocal and acoustic-guitar recordings.

We suggested creating an area where vocals could be recorded effectively by fitting a curtain rail around one corner and then hanging a heavy drape or duvet on this. Our remaining piece of two-inch Auralex foam was stuck to the ceiling with the idea that the singer would stand under that with their back to the drapes to record. The rest of the room wasn't too crucial, as the V-Drums don't care about acoustics and any guitar amps would be close-miked. Acoustic guitars could be recorded satisfactorily in the vocal position. We didn't have the time or fittings to put up the curtain rail, but Brett said he could arrange that with no problem.



Another foam panel was placed on the rear wall of the studio, along with a polystyrene diffuser designed to combat the high-frequency absorption of the carpeted studio walls.

It turned out that Brett's monitoring difficulties arose when he fed the singer a headphone mix from his MOTU 828 MkII interface and then tried to juggle the headphone and speaker monitoring level using its front-panel controls. The MOTU 828 MkII isn't really designed to do this, and both Hugh and I felt that the only satisfactory solution would be to use a monitor controller that allowed the cue headphone and speaker levels to be controlled independently. Furthermore, some monitor controllers have two headphone outputs, each with its own level control, so the vocalist can have one feed and Brett the other while tracking. This would allow him to track vocals with the monitors turned right down, thus avoiding spill into the live room. An additional benefit with a monitor controller of this type is that there's a talkback facility built in so you don't have to yell through the control-room glass. Brett agreed that this approach made a lot of sense and put in an order for a Mackie Big Knob the same week.

Troubleshooting Mackie Control

That left the Mackie Control to deal with. Brett's troubles with this had started when he bought two MIDI leads that were such a tight fit in the Mackie Control's sockets that he was almost afraid to pull them out again! I took a couple of MIDI cables along with me and they fitted fine, so I think he was probably unlucky and just happened to get a pair where the pins had been too heavily plated. Then again, this isn't the first time Brett has been unlucky with leads — when he bought his MOTU 828 MkII he couldn't get that going for ages, and the culprit turned out to be a faulty Firewire cable!

Once we'd switched on the Mackie Control, it was clear that it hadn't yet been set up to work as a Logic Control, but as I was unfamiliar with the unit I decided to break the habit of a lifetime and look in the manual. This showed which two buttons to hold down while turning the unit on to get it into the mode where you could choose what controller you wanted it to emulate. Pressing down the selection buttons on the first two channels entered the emulation mode, and then pressing the V-Pot on channel eight selected the Logic Control option, after which I configured it in *Logic's* Preferences.



The final remaining foam panel was placed on the ceiling in the live room, above where Brett was planning to set up his mic for vocal recording.

All seemed well, with the transport controls and faders behaving as they should, but a few more tests showed that we still had problems. The Jog wheel wouldn't jog and the pan controls would only pan to the extreme left or right with nothing in between. No amount of resetting or trashing *Logic's* Preferences would fix it, so I came to the conclusion that the most likely cause was a hardware or firmware fault, and the only way to prove that was with a replacement Mackie Control.

As Brett was a relative newcomer to *Logic*, I took some time to set up a default song for him with 24 audio tracks and 16 instrument tracks, which I thought would be more than enough for most of his projects. I showed him how to create screensets, then explained that the best way to deploy reverb plug-ins is to use them in busses, fed from post-fade aux sends, as that way all the channels can have access to the same effects.

If you simply drop a reverb plug-in into the insert point of every channel that needs reverb, you'll soon run out of CPU power, especially if you're using a power-hungry convolution reverb such as *Space Designer*. For most mixes two different reverbs will provide all the variety that's needed. Hugh and I also went through the basics of compression, as this is one area that a lot of people have

trouble with.

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Summing Up

By the middle of the afternoon the studio was sounding much more neutral, Brett had a good idea what needed adding to his gear list and, aside from the Mackie Control problems, his system was up and running, requiring only a suitable duvet before he could begin making first-rate vocal recordings. Once again, it is demonstrable that, while it may look very nice, the overuse of carpet as an acoustic treatment can really mess up your room acoustics unless balanced by some high-frequency reflection and low-frequency trapping. Given the choice, I wouldn't have used much if any in this room, but as we were faced with a finished job we had to incorporate what was already there into the final design.

The Auralex panels mounted on blocks really seemed to do the trick in sorting out the lower mid-range, though Brett's room would benefit from more bass trapping, and a large void under his desk area was still acting as a resonator to some extent, as it was an empty space enclosed on all sides but one. Filling this with mineral wool, foam, or even old bedding would help damp it down and provide low-cost bass trapping. **SOS**

Published in SOS August 2005

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Studio SOS

Jazz and Alessia

Published in SOS July 2005

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People : Studio SOS

The SOS team return to Cambridge, turning another home studio upside down in search of improved vocal and guitar sounds.

Paul White

Jazz and Alessia met while on an SAE course in Glasgow, and have now set up their own home studio in Cambridge to enable them to carry on composing and producing. They've already had some success in this field while studying in Glasgow — Jazz and his former production outfit had the same management company as Ms Dynamite, and did both composing and remixing work for the company. The work included remixes/work for Ms Dynamite, 50 Cent, Lisa Mafia, Da Endz, Working Title (for the film 'The Guru'), Lamya, and various Bhangra artists. Alessia writes, sings, and co-produces with Jazz. The pair are currently finishing their own album and hope to produce other artists as well as themselves in their own project studio.



Paul surveys Jazz and Alessia's home studio setup, and listens to some test tracks over the monitors to assess its performance.

The studio is set up in a first-floor room that is relatively long and narrow. After Jazz contacted us asking for help he sent over some photos of the room, and Technical Editor Hugh Robjohns and I both suspected that there might be monitoring problems to address. However, Jazz specifically wanted help with his vocal recording and mixing.

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Bass Inconsistency

When we arrived, the desk supporting the computer monitor and a new pair of ADAM P11A active speakers was set up approximately halfway along the length of the room, the idea being to divide the room into a control-room area and an area for recording vocals. Jazz was toying with the idea of building a vocal booth in part of the available performing space. The studio setup itself was pretty simple, with a Mac G5 running a Digidesign Digi 002 rack interface, and Ableton *Live* was used for creating and assembling ideas before mixing in *Pro Tools LE*. For processing, Jazz was using only the plug-ins that come with *Pro Tools LE*, though it turned out that these worked pretty well for the type of music he was working on.

As the ADAM monitors were new, Jazz wanted to check that they were performing correctly, but after playing a selection of material through them, it became evident that the level of bass dropped away rapidly if you moved back from the speakers, and the computer monitor was also set up slightly forward of the speakers, which can cause problems with reflections, particularly at high frequencies. In addition to the low-end consistency problem, the bass was not as well defined as it should have been and the stereo imaging was quite poor because of the proximity of bare plaster walls to either side.



Although the judicious application of some acoustic foam helped to clean up the stereo imaging somewhat, more drastic measures were clearly required for dealing with the room's uneven bass response.

Fortunately, we'd brought three large Auralex foam panels with us (supplied by the UK distributor Audio Agency) that we felt would improve the imaging if placed on the side walls. We also suggested that a pair of foam isolator pads under the monitors would improve the solidarity of the bass end, given that the monitors were set up on a computer desk shelf rather than on rigid stands. We improvised a pair of these using the packing that the speakers came with and, even though they were not ideal for the purpose, the sound improved noticeably, so we arranged for a pair of Auralex's MoPad speaker supports to be sent on later so that Jazz could evaluate the final sound.

While our little fixes improved the general sound and imaging of the monitoring quite noticeably, the unreliable bass variation with listening position wasn't so easy to fix — we've found that most small rooms have a Bermuda Triangle (well, a Bermuda sphere really!) in the centre where bass sinks without trace if you mix from there, and if you put your monitors there too things get even more uncertain.

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Room Reorganisation

After talking about what Jazz and Alessia hoped to get out of the room, we persuaded them to try moving everything around so the monitor desk was at one end of the room, facing the wall. In long narrow rooms, it is invariably better to fire the speakers down the length of the room rather than across it, as the reflections from the close wall behind the engineer tend to be very difficult to deal with when working across the room.

We spent a few minutes removing everything from the bookshelves currently occupying the end of the room where we wanted to set up the desk, then began clearing that half of the room ready to set up the studio in its new location. I even vacuumed the floor before we moved the desk — this is getting to be a habit — while Alessia polished the bare cupboards. To reduce the risk of ground-loop hum, we used a 'star' mains system, which is to say we plugged a distribution board into one mains socket, then ran other distribution boards from this so that everything was powered from the same place. Hugh taped one of the distribution boards to the rear support of the computer desk to get the wiring off the floor and to give me yet another excuse to photograph him clambering under furniture!



After some discussion it was decided that the best way to tackle the acoustic and operational problems Jazz and Alessia were encountering would be to reorganise the layout of the studio. This involved moving the shelves from the end wall, allowing the main studio workstation to be set up in that position instead.

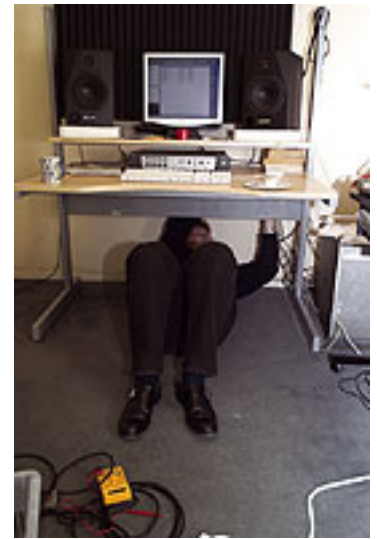
Jazz re-connected the G5 and the Digi 002 as we tried to organise the rest of the room. In a very short space of time, we had a much more spacious setup with all the necessary gear within easy reach of the engineer's seat and with plenty of clear floor space for recording vocals and guitars. During this process, the impressive mound of top-quality chocolate biscuits provided by Alessia diminished greatly!

The studio's car-boot-sale Hammond organ (a bargain at £10) was repositioned to the other side of the room, and the bookshelves were moved to the back of the room, against the side wall close to the bay window. A new purpose-designed table was being built to support Jazz's turntables and DJ mixer, and it would sit neatly between the bookcases and the keyboard rack.

As the house belongs to Alessia's father, I suggested that we prop the Auralex foam up in a temporary fashion to find the best place for it, then Jazz could procure some quarter-inch MDF panels, stick the Auralex to that, and hang the panels up like pictures using hooks. This would save the mess that invariably ensues if you ever need to remove foam that has been stuck directly to the wall. Either spray carpet adhesive (as supplied by Auralex) or a Liquid Nails-type adhesive works well for this.

Both Jazz and Alessia thought this was a good idea, so we proceeded on that basis, putting one panel on each side wall with the vertical centre of the panel at about seated head height, and the horizontal centre of the panel just forward of the listening position. Fortunately a radiator on the left wall gave us a temporary mounting option in about the right place, while a shelf unit on the right-hand side was almost as obliging. The third panel was propped up centrally on the wall behind the speakers using the desk as a support.

Playing our test songs now confirmed that we had a much more even bass response as we moved back from the normal listening position, and the bass was also more solid and far less 'tubby' sounding. The foam at either side brought better focus to the stereo imaging as well. Jazz mentioned getting some foam bass traps, and we agreed this would be a good idea — he'd probably want to place them over the full height of the two front corners of the room.



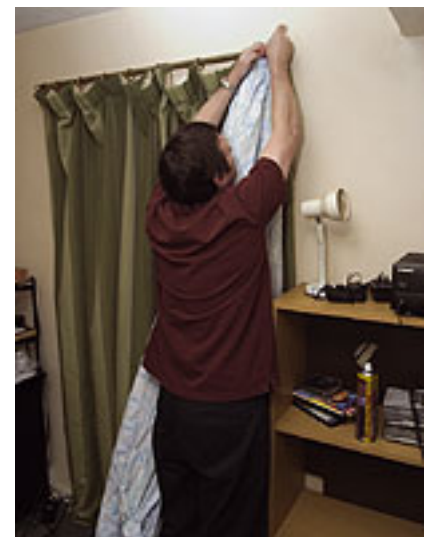
Once the main workstation was in its new position, Hugh set to work creating an efficient 'star' mains wiring scheme. This included fixing mains distribution boards to one of the main desk's supports, allowing the mains cables to be tidied off the floor.

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Vocal Recording Advice

With the monitoring now performing far better, we felt we could move on to the vocal-recording problems. The couple had recently bought a Neumann TLM103 vocal condenser mic, which we set up in its shockmount with a pop shield positioned a couple of inches in front of it. Alessia volunteered to do some singing so we could check our results, but before doing that we loaded in a couple of songs she'd already been working on to see what the vocal was like when heard in isolation.

Both Hugh and I were extremely impressed with Alessia's contemporary vocal style, but we could hear some room coloration clouding the lower range of the vocal mic. There was also a tendency for the mic to sound as though it was on the verge of popping sometimes. Alessia often records vocals when Jazz is at work, and she usually hangs a double duvet off the curtain rail that runs across the bay window. Duvets can be great low-cost problem-solvers, but there were so many bare plaster surfaces on the other walls that some room coloration was still clearly audible.



An impromptu solution was to pull the curtain across the bay window (it was fairly heavy and absorbed sound reasonably well) and then to hang the duvet on the adjacent side wall by tying loops of string to the top two corners, then looping these over the ends of the bay-window and side-door curtain poles — these were almost ideally placed for this purpose. We now had a corner that was reasonably dead on two sides, so if the vocals are recorded with the mic facing into the corner, most of the rogue reflections should be intercepted before getting into the mic (which has a cardioid polar pattern). Using a thicker duvet over the bay window curtain might also help. There was space on a roof beam to hang yet another double duvet to act as a screen between the booth and control room areas, and which would effectively form a three-sided vocal booth. Alessia also suggested hanging another duvet over the door to hide the reflective wood surface — every bit helps!

We felt that working like this would be better than trying to build a separate vocal booth, as these nearly always sound boxy and are quite difficult to treat in an acoustically neutral way.

To test our 'one duvet and adjacent curtain' setup, Jazz opened up a song they'd been working on and Alessia overdubbed a new vocal verse. This was also to serve as the test-bed for any processing we decided to try. She worked about six inches from the pop shield with her back facing the duvet. Just one take and we had a perfectly respectable piece of test vocal. We never did figure out why she was getting traces of popping before, as she always used the pop shield and also the bass-cut switch on the 002 preamp, although it was possible that the pop shield had been arranged too close to the mic itself — there should be a gap of a couple of inches of still air between the shield and mic. However, with our setup, we didn't get any hint of popping at all.

Listening back to what we'd recorded, we noticed straightaway that the room coloration was much reduced and that we had a good basic tone to work with before using EQ or other processing. Jazz had been using the *Pro Tools* reverb, but I felt a much shorter, brighter setting than the one he'd been using would better suit both the song and Alessia's voice. I hadn't used the *Pro Tools LE* plug-in reverb before, so I was interested to see what I could get out of it. I ended up using the medium plate with the decay time turned right down to 650ms and with a 70ms pre-delay. I left the filters at a fairly bright setting and turned the diffusion down to 45 percent to try to coax something closer to an ambience effect out of it. I didn't want an obvious reverb, but the vocal needed a sense of life and space.



With the studio workstation set up at one end of the room, a corner at the other end was now set up as a vocal recording area. In order to reduce the amount of room sound reaching the vocal mic, Paul closed the heavy bay-window curtain and also deadened the wall by hanging a double duvet between the two windows — the existing curtain rails made convenient fixing points.

This combination of parameters worked astonishingly well, and both Jazz and Alessia really liked the result. Jazz had been considering buying and using an outboard reverb, so we patched in a Lexicon MPX550 that I'd brought with me and set it up to give a similar sound to the plug-in reverb. None of us felt that it was any better in this application — which pleased Jazz, as he didn't want to have to budget for an outboard reverb if he could help it.

Although the new room arrangement and duvet hanging had removed most of the room coloration, there was the mildest hint of boxiness to the vocal sound which we tamed with a very gentle 200Hz dip using the *Pro Tools LE* four-band EQ plug-in. I also added a little 'fairy dust' boost centred at 10kHz to give a hint of 'air'. The amounts of EQ gain involved were only 2-3dB, but the vocal clarity improved and it felt much more upfront in the mix, which is what Jazz and Alessia wanted to achieve.

The final tweak was to add some compression. Alessia has really good mic technique, so compression wasn't needed to keep the level even at all. Instead, we were able to use it to add a little 'punch' and to create a more contemporary sound. We settled on a hard-knee, 8:1-ratio compression with an attack time of 3.7ms and a 100ms release time. The threshold was adjusted to give a maximum gain reduction of around 6dB on peaks and it was working pretty much all the time Alessia was singing.

I felt the result sounded really crisp and solid, and as a further demonstration of compression as an effect, I pushed the gain reduction up to around 15 to 20dB (by reducing the threshold) to show how compression could be used to produce a more assertive sound — almost like singing through a limiter. Of course, compression brings up background noise, breath sounds, and room ambience between phrases, so you really need good acoustics and a quiet room to get the best out of this technique. Using a gate or expander on the vocals prior to compression can also help.

Jazz then asked about panning



The reorganised studio layout instantly improved the evenness of the bass response, and a quick test vocal take indicated that the vocal sound was also much improved.



Once the test vocal had been recorded, Paul worked with Jazz and Alessia on refining a usable vocal ambience setting using the built-in plug-ins within *Pro Tools LE*. This worked so well that no appreciable improvement in quality could be achieved even using a dedicated external Lexicon MPX550 reverb unit.

strategies for backing vocals, as Alessia likes to layer a considerable number of different harmony parts. Hugh pointed out that extreme panning can lead to perceived balance problems when the mix is heard in mono, so we suggested keeping the main vocals in the centre as usual, panning the main backing vocals to around 10 o'clock and two o'clock, and saving extreme panning for any extra backing vocals that wouldn't cause the overall mix to suffer too much if they lost a little level when played back in mono. It's also possible to add space to a multi-layer backing-vocal mix by rolling out all the low end, say below 200Hz, and adding more reverb to those parts that are panned towards the edges. This retains transparency in the mix and also adds a nice gloss to the production.

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Miking Classical Guitar

Our final task was to try miking Alessia's Encore nylon-strung classical guitar. This turned out to be very easy, as classical guitars tend not to boom in quite the same way as metal-strung models, so you don't have to worry as much about aiming the mic too close to the soundhole. All acoustic-guitar recordings seem to benefit from some early reflections, either from a hard wooden floor, or from some kind of 'acoustic mirror'. The only suitable sized reflecting board to hand was one of the computer desk's side shelves, but this served the purpose when supported on a handy cable box and angled to bounce some of the guitar's sound back into the mic.



In order to improve Jazz and Alessia's acoustic guitar recordings, Paul set up an acoustic reflector (otherwise known as a spare desk shelf!) to liven up the sound reaching the mic. Paul then encouraged Jazz to move the mic around while listening to his playing on headphones, so that he could hear how much difference small position changes can make.

Jazz was considering replacing the studio carpet with laminate flooring, but I felt this would be a disadvantage, as the noise from the computer would become more evident with nothing to soak it up. A reflective floor isn't always great for vocals either, as it can colour the sound. The compromise solution is either to floor only a small section at the end of the room for recording acoustic guitars, or simply to put a wooden panel over the carpet when recording those instruments that need a bit of reflective help.

While wearing headphones, I moved the guitar relative to the mic until we found a sweet spot that came close to the natural acoustic sound of the instrument. I encouraged Jazz to move the mic around while listening on headphones to hear for himself how moving the mic small distances dramatically changes the recorded sound. In this case, the best result turned out to be achieved by aiming the mic from 12-18 inches away at a point midway between the end of the neck and the soundhole.

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Jazz & Alessia's Comments

Moving the furniture around has spread the bass evenly throughout the studio, and hanging the acoustic foam on the walls has also helped to give a clearer and wider stereo image. Another welcome side-effect of the reorganisation is that the room now appears larger and more inviting! Our vocal recordings are much dryer with a lot less coloration thanks to the correct mic positioning and duvet padding. It has made a huge difference overall — thank you SOS!

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Lessons Learned

Although moving the whole studio around and re-plugging everything took a little time, I feel it really paid off in terms of more accurate monitoring and better use of the available space. The simple Auralex treatment took care of the worst local reflections around the listening position, and while our improvised speaker pads weren't great, they helped too. Switching these for the Auralex MoPads when they arrive should tighten up the low end even more, and adding bass traps to the room corners will further improve the evenness of the bass.

Jazz said that he might get a second computer monitor, in which case there wouldn't be room on his existing speaker shelf for both monitors and both speakers. One solution would be to cut off the metal vertical supports protruding above the current monitor shelf, fitting a wider shelf on top, but a more acoustically sound solution might be to use rigid speaker stands either side of the computer desk (provided that these don't place the monitors closer than around one foot from the side walls). Flat screen monitors aren't affected by the magnets in monitor speakers, so you can put them as close together as you like.



The simple duvet treatments really worked well for the vocal recording area, and adding more duvets, as suggested by Alessia, should improve things further. As stated earlier, this will almost certainly lead to better results than offered by a small, enclosed vocal booth. Our final experiments with processing confirmed that Alessia's already excellent contemporary voice could be made to sit better in the mix with just a little EQ, compression, and a short, bright reverb. Too much reverb can swamp a voice, but the treatment we chose actually helped pull it to the front of the mix. The final challenge before hitting the road and heading back to the Midlands was to eat a portion of Alessia's home-made pasta without coming back for seconds. We failed! **SOS**

Published in SOS July 2005

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Studio SOS

Rod Brakes

Published in SOS June 2005

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People : Studio SOS

The owner of an unusually bijou studio setup provides the chocolate biscuits this month, as the SOS team get busy helping to improve the performance of his gear and the sound of his recordings.

Paul White

Rod Brakes' studio is probably the smallest the Studio SOS team has ever seen, but he still has everything necessary to produce good-quality recordings. His main instrument is the guitar and his home studio setup includes several shelves full of pedals and processors dedicated to creating new and unusual guitar sounds.

Space is clearly a limitation, and the studio is set up as a booth in one corner of his living room, where a set of pine shelves forms a partition to his left and provides space for his computer, as well as for some storage (see photo on page 102). The space between the wall on the right and the shelves on the left is little over a metre, so this clearly isn't the ideal place to set up accurate monitoring. Rod called us in mainly to concentrate on his acoustic and electric guitar recording technique but, as always, we tried to improve the monitoring first.

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Hear Here: Monitoring

Working in such a confined space, Rod had wisely chosen some fairly small



The lovely Martin acoustic wasn't sounding as good as Rod Brakes had hoped when recorded, so Paul threw on some headphones and set to locating a miking position for the Rode NT1000 that would truly reflect the sound of the guitar as Rod experienced it while playing.

monitors, in this case the surprisingly effective and inexpensive Egosys Near 05 Active 129 models he'd picked up from Turnkey. When we arrived, these were mounted directly on his glass-topped desk — along with his flat-screen computer monitor, outboard rack and QWERTY keyboard — but they were facing backwards! However, this wasn't some weird new monitoring technique: Rod has a very young daughter with a propensity for sticking her hands into exposed tweeters, so he turns the monitors around to face the rear wall when they're not in use, to protect them.

As Rod had them set up on his desk, the monitors were much too low, so we raised them slightly on a set of Auralex foam speaker pads, configured back-to-front to allow the speakers to be angled upwards (rather than down). The extra foam wedges provided with the pads were also used to increase the upward angle, which resulted in the tweeters firing more or less directly towards the listener's head — exactly as it should be. UK Auralex distributor, Audio Agency, kindly donated the speaker pads to the cause, along with a few panels of acoustic foam.

Playing our test audio disc showed that the monitoring actually sounded a lot better than we'd expected in such a tight location, but the stereo image was somewhat messed up by reflections from the nearby wall on the right and the computer on the left. We improved this situation by fixing just two Auralex 24-inch-square foam tiles to the left and right of the mixing position at head height, one glued to the wall (using a spray carpet adhesive from my local DIY store costing £3.99 a can) and the other wedged temporarily next to the computer on the pine shelves. Rod planned to use map pins to fix this more securely to the side of the shelving unit. The result was a slightly tighter sound with noticeably better stereo imaging. A further benefit was that the foam on the left also reduced noise from the computer. Our verdict was that, with care, Rod could achieve decent mixes with this simple setup, but that if he intended to release any of it he'd ideally need to get it mastered somewhere that had accurate monitoring.

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Rod's System & Methods

The rest of Rod's system is based around Steinberg *Cubase VST* software running on a fairly brisk PC with an Emu 1820M audio interface. Level control for the monitoring is done in the Emu software that essentially channels audio into and out of *Cubase VST*, so one of our first recommendations was that Rod buy a simple monitor controller, such as the Samson C-Control, to allow him hands-on control over monitoring level and also to let him play his CDs or keyboards without first having to fire up the computer. This same box would also provide a headphone output and talkback. Talkback might not seem such a big deal in such a small studio, but Rod tends to use different rooms for recording different instruments, the bathroom being a favourite for acoustic guitar. Having talkback would enable him to communicate with the player in the bathroom via the headphones, although most of Rod's current projects seem to involve him doing all the playing, singing and engineering.

Most people recording an electric guitar in a small domestic studio would use something like a POD XT or other direct-recording device for convenience and flexibility, but Rod is a bit of a traditionalist and has a 100W Marshall JMP Mk II Lead amplifier (dating from 1979) feeding a Marshall 4x12 cabinet built in the '60s — one of the ones fitted with Sutton Surrey Celestion 'green back' G12M speakers. This isn't quite as rash as it might seem, though, as Rod houses the cabinet in a kitchen closet facing a wall covered with acoustic foam and runs cables under the door. There's just about room between the speaker and the wall to get a microphone into position and that's how he records the cabinet sound.



Rod's monitors were far too low on the desk. Raising them and angling them upwards slightly on foam pads made a big improvement to the monitoring sound.

His traditionalist approach also extends to synths and drum machines. He has just one soft-synth bundle but is most happy when he's putting the separate outputs from his cherished Roland TR909 drum machine through his guitar pedals to wring something new out of it. His only hardware synth is a three-octave Roland Sound Canvas keyboard.

Rod has been DI'ing his Ibanez bass guitar via an Electro-Harmonix Black Finger optical tube compressor and a little Dbx Vacuum Tube Preamp, then feeding the output from this into the mic input of his Emu interface. This particular Dbx preamp has a high input impedance on the line jack, and so is ideal for guitar and bass DI purposes. However, we suggested he used the line input of the Emu interface in future, as the levels would be better matched and the result would be less noise in his recordings.

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Tweaking The Drums

Before trying anything else, we sat down to listen to some of Rod's own recordings while he furnished us with cappuccino and Hobnobs. (If anyone else is thinking of asking us over, might we mention that there is now a new Extra Chocolate version of Hobnobs that we haven't tried yet?) As it turned out, Rod's recordings were all pretty clean and tight-sounding, and his traditional approach even extended to recording his own drum samples from an acoustic kit, with drums recorded individually using a Shure Beta 57 dynamic mic and a Rode NT1000 capacitor mic. The only serious flaw was the sound of the kick-drum sample, which was quite short and 'boxy'. Kick drums really need a dedicated mic with lots of low end to record them convincingly, and although I knew that processing was never going to make this one sound great I decided to have a go anyway, using the EQ and compressor in *Cubase VST*.

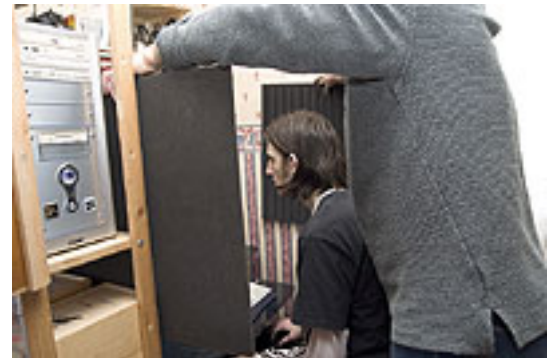
To improve the overall sound, I set up an EQ with a fairly aggressive boost at 90Hz, a dip at around 200Hz and another peak in the 4.5kHz region, the idea being to lose some of the boxiness, beef up the low end and emphasise the click of the beater. There was a very noticeable improvement in tone, and after I compressed the kick sound with auto release and an attack of 10ms, then switched on the soft-clip limiter, the kick sat reasonably well in the mix — although it still wasn't going to win prizes. Rod's best bet is to become slightly less purist and substitute a nice bass-drum sample from elsewhere!

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Getting To Grips With Guitar

Turning our attention to the rest of the mix, it seemed that Rod had tried to optimise each sound in isolation, the result being that when they were playing together they all fought to be at the front of the mix. This wasn't so bad in the intro, where only a couple of instruments were playing, but it got fairly messy when the electric guitars came in. The acoustic guitar, though clear, had a somewhat hard quality to it and the DI'd bass also sounded quite assertive. I tried some EQ on the acoustic guitar to cut the region between 500Hz and 1kHz, which took some of the hardness out of it, but the real solution to these problems was to examine Rod's recording technique and see what could be done to improve the sounds at source.

Acoustic guitar figures quite highly in Rod's music and he'd recently treated himself to a mahogany Martin Auditorium guitar, which sounded rather nice played acoustically. Rod also uses alternate tunings, such as DADGAD, which suit this guitar well. Normally he records the guitar using his Rode NT1000 (his only capacitor mic) or his Beta 57, with the common technique of pointing the mic towards the position where the neck meets the body. He generally uses the bathroom to add a bit of life to the sound, but we decided to experiment in the living room, where there was more space, then move to the bathroom after we'd established a working method.



The trusty foam came out again to improve the sound of Rod's listening position. Here Paul's holding two foam tiles in place so that Rod can hear the difference they make.

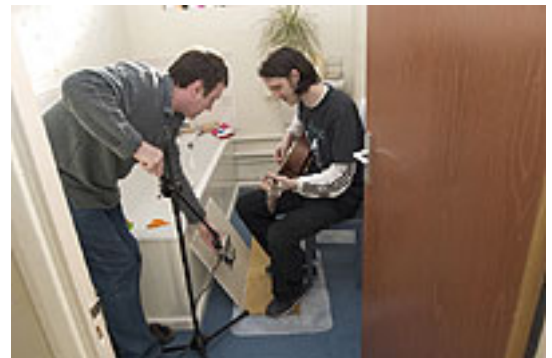
Given the choice, I'd never record an acoustic guitar using a dynamic mic, other than perhaps a Sennheiser MD441, as capacitor mics generally handle the top end more effectively and they're also more sensitive, which helps keep noise down. So we rigged the Rode NT1000, initially with the mic in Rod's usual position (aiming at the neck/body joint from 12 to 18 inches in front of the guitar). We also put a couple of boards on the floor in front of Rod, to reflect some sound back up into the mic. The board furthest from him was angled up, originally

because a chair was in the way, but it sounded good so we left it! We used boards because the living room was carpeted and acoustic guitars often respond well to reflective wooden or tiled floors (see photo on the first page of this feature).

With Hugh Robjohns driving *Cubase* we made some test recordings and, sure enough, the result was the same clear but hard sound that Rod had been getting. Apparently, Rod had also tried the higher mic position he'd read about in one of my books, where the mic, in effect, looks over the player's right shoulder, but it still sounded too hard. This just goes to confirm that the sweet spot varies enormously, depending on the characteristics of both the mic and the guitar, plus the tonal preferences of the player or engineer.

The task was now to try to find a mic location that would give a sound close to what we felt we were hearing acoustically from the instrument, and I did this by monitoring via headphones while moving the mic around. Rod played as I moved the mic, and eventually I found a great-sounding spot around a foot from the floor and two feet from the face of the guitar, with the mic aimed up at the 12th fret. Here the sound was really sweet and airy, without any of the harshness we'd heard before. After making more test recordings, we played the original and new mic-placement versions back to Rod and he agreed that the new placement had really captured the qualities of the guitar that he heard while playing it.

Transferring this setup to the bathroom gave us a similar tone, overlaid with a short, bright room ambience, but we had to move Rod around so that he wasn't playing parallel to the wall, in order to make the sound suitably subtle. At this point, Rod asked if it was worth miking the guitar in stereo. This is largely a matter of taste. My own view is that mono miking works well where the guitar is part of a busy mix, but stereo can add dimension to the sound where the guitar is a featured instrument. I'd brought along my own SE Electronics SE1a small-diaphragm cardioid mic, in case extra mics were needed, but you can try this technique with just about any decent capacitor model.



Having worked out how to mike the guitar for a good sound in the lounge, Paul and Rod adjourned to the bathroom, where Rod likes to make use of the ambience, to settle on the best seating and miking positions there. Rod also wanted to try miking in stereo (lower photo).

Rather than using a traditional spaced-pair or coincident-pair approach, we left the NT1000 where it was, as the main mic, and set up the SE1a on a separate stand around six inches from the headstock of the guitar, aiming it more or less at the nut (see photos on page 103). More test recordings confirmed that this produced a very different and much brighter tone than the main NT1000. With equal contributions from both mics the sound was probably too bright for most applications, but by reducing the contribution from the mic on the headstock and panning the two sounds apart, we were able to achieve a very nice, spacious sound with a good tone. In this type of situation, the room ambience can be regulated quite effectively by hanging towels over the bath and radiators, and for this session we found that a towel over the radiator definitely helped. The final sound had just the right amount of ambience to allow it to work in a mix with little or no added reverb.

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Rod's Comments On The Session

"It's not every day you get the chance for two of the world's most respected experts on music technology to visit your home and spend the afternoon giving you a one-to-one home-studio consultation! I was and still am blown away. I will never forget it. Thanks guys.



"Now everything sounds infinitely better. I certainly don't take the Auralex acoustic foam gifts lightly and I really appreciate them. They make a massive difference to the way things sound.

"As well as being great company, Paul and Hugh were both amazingly helpful and insightful; I feel as though I've learned a huge amount following the visit, and have been immediately putting all the hints and tips into practice. For example, I've just recorded a superb take of an instrumental acoustic number that sounds absolutely first-rate thanks to Paul and Hugh finding the 'sweet spot' for my Martin, angling the mic correctly and making use of some reflective surfaces and ambience. It sounds so good from source now; this one's gonna need pretty much sweet FA in terms of further processing. Nothing short of a triumph!

"Overall, Paul and Hugh's comments were very encouraging, and all I can say is that I must be paying attention when I read Paul's books and SOS! Now all that remains is for me to record a great album..."

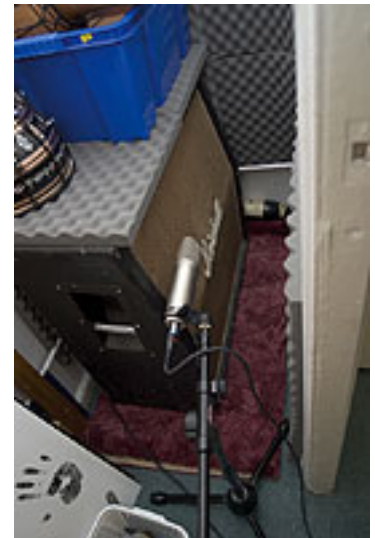
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Improving The Electric

Our next port of call, after the ginger cookies had been consumed, was the electric guitar. I brought along my POD XT for comparison and Rod was impressed both with its versatility and its ability to emulate his miked Marshall fairly authentically. We also coaxed a very nice DI'd bass sound out of it with little

effort. However, as he was already getting the sounds he wanted from his Marshall, he felt that sticking with that would be more organic, especially as he'd built up an impressive collection of Electro-Harmonix pedals and other processors to use with it.

We decided to work through the guitar-cab miking method to see if we could improve on it, and I achieved what I felt was an accurate interpretation of the amp sound by using the NT1000 about two inches away from one of the upper speakers. Rod had been miking one of the lower speakers, but we thought it was a good idea to move the mic up, to minimise any reflection effects from the floor. The mic was positioned roughly midway between the centre and edge of the cone, which worked very well, with surprisingly little sound leakage back through to the studio. However, I felt that the mid-range might benefit if the thickness of foam on the wall behind the mic was doubled, and as we had one piece of foam remaining we left it with Rod to install in the cupboard himself.



Rod prefers the traditional approach to electric guitar sounds, miking up a Marshall cab that's hidden away in a cupboard.

Rod then set about demonstrating some of the sounds he likes, using his processors in front of the Marshall, which he generally leaves set to a fairly clean sound. He has an Electro-Harmonix Hot Tubes overdrive pedal, a Marshall Guv'nor distortion pedal, Boss DS2 Turbo Distortion and OS2 Overdrive/ Distortion pedals for producing overdrive, and he's customised his Mexican Fender Strat by fitting a Deluxe-style roller nut and locking Sperzel machine-heads. He's also changed two of the pickups. The bridge pickup is now a Seymour Duncan Hot Rail and the neck pickup a Quarter Pounder from the same company. A non-locking mute button has also been installed for creating effects.

For special effects, Rod has an E-bow, which he was still coming to grips with at the time of our visit, but his *pièce de resistance* is using wah, overdrive and echo while playing the guitar with what might be most safely described as a battery-operated, variable-speed personal massaging device! The golden vibrating tip is effective in exciting the strings, while the electromagnetic interaction between the varispeed motor and the guitar pickups produces a high-energy sound not unlike a light sabre burning its way through a Morris Minor engine block! Apparently, he got some very funny looks while in the shop trying to select the most appropriate model...

Our slightly tweaked mic setup was delivering the goods with the guitar sound, but one thing I noticed was that the Marshall amp had a slightly 'barky' quality when its own overdrive was invoked, which suggests that the output valves may need to be checked, along with their biasing.

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Which Sample Rate?

At the end of the session, Rod asked us whether he should be working at 44.1kHz or 48kHz, as he felt 48kHz would give a better result. Technically, 48kHz can sound a little better than 44.1kHz, but in a system like this one there is unlikely to be any detectable difference — and by the time a 48kHz recording has been reduced to 44.1kHz, by means of the *Cubase* sample-rate converter, it may actually sound less good than a straight 44.1kHz recording. Both Hugh and I felt that working at 44.1kHz and 24-bit would be best, leaving dithering to 16-bit as the last step of mastering.

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Optimising The Processors

The final part of our assignment was to advise on the best way for Rod to use his Aphex 204 Big Bottom Exciter, his Focusrite Compounder and his new Lexicon MPX550 reverb unit. These sit in a rack on the desk beneath his computer monitor, and previously he'd passed the entire mix through all three in series. This is, of course, quite limiting, as the same amount of processing is applied to everything, but there is a limit as to how flexibly you can use hardware when your audio interface has only eight inputs and outputs.

Our final recommendation was to set up an aux send and return in *Cubase* VST (using a spare output and two inputs on the Emu interface) and use this to feed the MPX550, set to 100 percent wet. In this way, differing amounts of reverb can be added to each of the VST tracks and the quality will be much better than the rather indifferent software reverbs that come with *Cubase* VST. Furthermore, because the MPX550 has digital I/O, it can be connected via the S/PDIF socket of any suitable audio interface that has S/PDIF I/O, to avoid an extra stage of analogue-to-digital conversion. The Compounder and Exciter can be left at the output of the chain to process the stereo mix, which can then be recorded back into the system as a new stereo track. Used with care, these two processors are very good for overall mix sweetening, while the EQ and dynamics in *Cubase* are perfectly adequate for most track-tweaking needs.



Er... actually, we said show us your *vibrato*...

Our final advice to Rod was to burn test CDs and play them on as many systems as possible, to allow him to get used to and compensate for the inaccuracies of his monitoring system. Though it's not a bad monitoring setup, the bass end will inevitably be misleading to a greater or lesser degree with such small speakers in such a compact location. **SOS**

|Published in SOS June 2005

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SOUND ON SOUND

The World's Best Music Recording Magazine

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Studio SOS

East Norfolk Sixth Form College

Published in SOS May 2005

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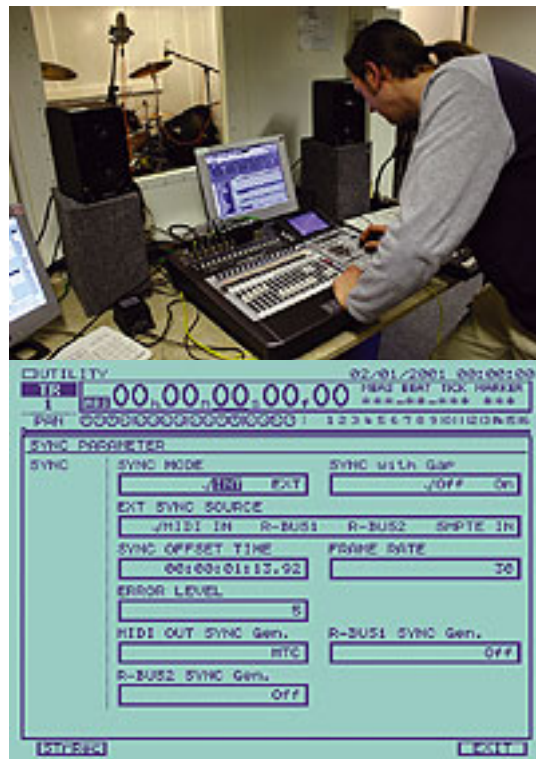
People : Studio SOS

East Norfolk Sixth Form College needed help integrating their hardware multitracker with their computer sequencing system, so the SOS team travelled over to Great Yarmouth to lend a hand.

Mike Senior

Back in the January and February issues of SOS I wrote a pair of workshop articles about the Roland VS2480 recorder. Within a couple of weeks of the second article going to press, I'd had two separate reader inquiries at the SOS office asking for a tutorial on the multitracker's remote control functions. Having a dedicated programmable fader bank in my own setup, I'd never needed to experiment with this aspect of the VS2480's operation, so I couldn't really provide any useful hands-on advice. However, I knew how the remote control functions were meant to work in theory, so I offered to head over to the studio of one of the two readers, Carl Simmonds, to help him get things worked out in practice.

Carl teaches music technology at East Norfolk Sixth Form College, where he manages a studio setup based around a Triton LE keyboard workstation, a PC running Steinberg *Cubase SX*, and a



Roland VS2480 recorder. There were two main things which Carl wanted to do. Firstly, he wanted to configure the system such that the computer sequencer and the hardware multitracker would run in sync, under the direction of the hardware transport controls. Then he wanted to be able to use the VS2480's faders to control mixer parameters within *Cubase SX*. There is a dedicated fader layer within the Roland digital mixer for this purpose, but Carl was unsure how to get this working properly.

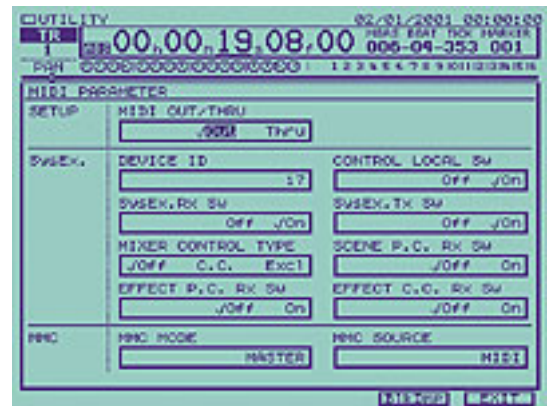
I enlisted the help of SOS contributor Tom Flint, himself an experienced user of digital multitrackers, and we headed over to Great Yarmouth where the college is based. On our arrival we found the small control room already half full of people, as some of the Music Technology 'A'-level students were keen to be in on the session, despite our visit coinciding with the half-term holidays!

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Synchronising Cubase SX To A Roland VS2480

With barely a pause to inhale our cups of tea, we set to work on synchronising the multitracker with the sequencer. The first thing to do was make sure that the multitracker was set up right. I explained to Carl that the most sensible method of synchronisation in his case would probably involve using MIDI Time Code (MTC). The other option would have been to use MIDI Clock, but that requires identical tempo and time-signature data to be set up in the sequencer and the multitracker, so can be a hassle if you have changes of time signature or tempo within any song. I also suggested that it would make most sense to have the sequencer as slave and the VS2480 as master, as most of the audio recording is done on the multitracker.

Dealing with the VS2480 was fairly quick — in the Utility menu the Sync page needed its MIDI Out Sync Gen parameter setting to MTC so that the multitracker would output MIDI Time Code, and I also checked that the MMC Mode switch in the MIDI settings page was at its default Master position. Transmission of SysEx messages is usually switched on by default, but I also checked this while I was in



Photos: Tom Flint

In order to synchronise the Roland VS2480 with *Cubase SX*, Mike first adjusted the Utility settings of the multitracker as shown in the insets (above): MIDI Out Sync Gen was set to MTC so that the VS2480 would send MIDI Time Code to the sequencer; MMC Mode was set to Master so that MIDI Machine Control messages would also be sent; and SysEx Tx Switch was set to On so that these messages would not be filtered out before reaching the multitracker's output. Finally, Mike hooked his own Yamaha QY700 (below) to the VS2480 to test that the setup was correct before starting work on *Cubase SX*.

the MIDI settings page. In my own studio I run a VS2480 synchronised to a Yamaha QY700 hardware sequencer, which I'd brought along so that we could check that the college's VS2480 was working correctly before we delved into *Cubase SX*. Having connected the multitracker's MIDI Out to the MIDI In of my QY700, the sequencer synchronised immediately, so we re-plugged to the studio computer's MIDI In and set about sorting *Cubase SX*.

Being more of an Apple *Logic* user myself, I asked Carl to navigate to *Cubase SX*'s Synchronisation Setup window, where we selected MIDI Time Code as the Timecode Source and specified the MIDI interface input port to which the VS2480's MIDI Out was connected. Finally, MIDI Machine Control was activated so that Cubase would respond to the VS2480's transport controls. Pressing Play on the VS2480 at this point had no effect on *Cubase SX*, so we had a quick look at the Transport bar and realised that the Sync button wasn't yet switched on. However, even switching that on yielded no result.



In order to get *Cubase SX* synchronising to the VS2480 (which was now sending MIDI Time Code and Machine Control messages), *Cubase SX*'s Synchronisation Setup window was configured as shown. In addition, the Sync button on the Transport panel was switched on (inset).

Knowing that we'd tested the VS2480 with the QY700, there was little doubt that the multitracker was set up correctly. Furthermore, MIDI was definitely reaching the sequencer, as could be seen from the MIDI input meter on *Cubase SX*'s Transport panel. This led us to suspect that we hadn't routed the MIDI Time Code correctly in the sequencer, so we tried setting different MIDI interface input ports from the Synchronisation Setup window. All of a sudden *Cubase SX* sprang into life, synchronising perfectly with the VS2480. When we'd first configured the Synchronisation Setup we'd set the wrong MIDI input port! It's at times like these that it pays off to have worked methodically — if we hadn't tested that the VS2480 was set up correctly before tangling with *Cubase SX* it could have taken us much longer to get to the crux of the problem.

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VS2480 Tips

While I was working on Carl's VS2480, I noticed a number of things which would help him work more efficiently. For example, he had paired the channels of very few of his stereo signals, because he wanted independent control over the individual pan parameters. However, you can still access individual pan settings for stereo-linked channels if you cursor over to the Pan control in the channel parameter screen and press Enter — this brings up a little window with the separate pan values and the communal balance.



Another thing which speeds up the mixing process is setting the Knob/Fader Assign Switch in the Utility menu's Global pages to its Fader setting. This means that you can easily view and adjust aux send levels across sixteen channels at once, which is very useful when you're trying to remember which channels are sending to which effects. I also showed Carl how the User Knob/Fader Assign mode can be used to transform the channel fader into an EQ bypass switch for when you're setting up the channel equalisers — after all, it's very good practice to keep switching the processing in and out of circuit while you're deciding on the right setting.

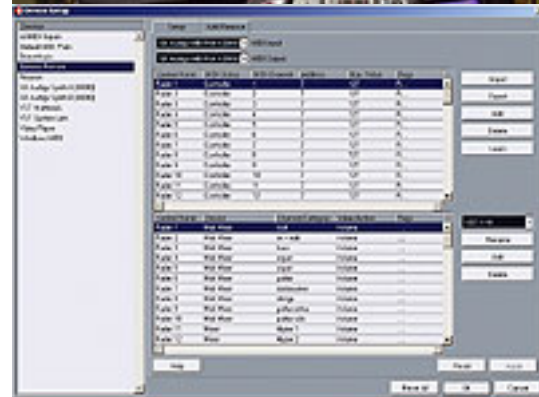
The multitracker's internal patchbay was another source of frustration. Carl had been having to keep resetting it for each new project, because the default template wasn't suitable for the college's setup. Fortunately, the VS2480 patchbay has an option to save routing templates, and I demonstrated that these could be shared between different projects to solve his problem.

Finally, I demonstrated how, when using the VGA monitor option, you can lock the unit's LCD to show the waveform display permanently — very useful if you do a lot of editing. First you have to press the Page button by the LCD until the IDWave option appears over the F3 button. This switches you to waveform view, and then pressing F6 (IDHold) locks this view to the LCD — however, this waveform display will always show the currently selected channel.

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V-Fader Control

The second task was to sort out the best way to control *Cubase SX*'s mixer from the VS2480's faders and rotary controls. The VS2480 has a dedicated fader layer (called V-Fader) within its digital mixer specifically designed for controlling external units. You enter the V-Fader mode by holding down the Shift key and pressing the bottom right-hand one of the Fader buttons above the master fader — the button has the V-Fader label underneath it. Each channel of the V-Fader layer sends out MIDI data on a separate MIDI channel, so the eighth fader and rotary control both send out MIDI on channel eight, for example. The controls can



Because no pre-programmed control-surface template was available for the VS2480 within *Cubase SX*, Carl and Mike decided to set up remote control from scratch using the Generic Remote profile in the Device Setup window.

file:///F:/SoS/SOS%20-%20Studio%20SOS%20Through%2010-2005/May%202005.htm (5 of 8)10/4/2005 12:22:45 AM

Knowing that the VS2480's audio-mixer faders can send *and* receive MIDI controllers, and that Carl was only using a handful of these for mixing the VS2480's analogue inputs, I decided to quickly try using these to control *Cubase SX* in place of the faders on the V-Fader layer. Unfortunately, the sequencer's MIDI Thru function caused the hardware faders to 'fight' me whenever I attempted to move them, because the *Cubase SX* audio Faders were re-transmitting every Continuous Controller message back to the VS2480 as they were received. In the end, Carl settled for controlling the *Cubase SX* mixer from the V-Fader layer — at least this allowed sensible hardware control for projects up until the point at which software mixer automation was used.

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Mixing Horns & Rhythm Guitars

Before we headed back, Carl played us one of the recordings that the students had made on the VS2480 — a funk-style track with drums, bass, guitars, and horns. The quality of the production was pretty impressive, with a particularly nice drum sound despite the main recording space being a 4 x 4m Esmono isolation room. This is a soundproof metal booth which can be built inside your own room to decrease sound leakage while recording. I encountered one of these rooms for the first time while interviewing Andy Cross for a Readerzone article back in *SOS* January 2003, and was struck not only by the effectiveness of the soundproofing, but also by how dry the internal acoustics were. The college's room was no exception — the room absorbed so much sound that you even felt yourself having to work hard to hold a conversation in there!

Admittedly, the drum sound was quite dry, but it suited the funk style of the track. I mentioned to Carl that he might have problems capturing any more roomy rock sound, and suggested that recording drum ambience from out in the corridor would probably help in such situations. I also suggested that he look at investing in one of the latest convolving reverb plug-ins — with such dry recordings he has to rely quite heavily on artificial reverb and delay, so I can imagine that high-quality reverb would make a real impact at the mixdown stage. Another alternative where a more natural recorded result was required would be to use hardboard panels to make the walls and floor more reflective when recording acoustic instruments.



The college's main recording area was a large Esmono isolation room. Although this provided great soundproofing, the internal acoustic was exceptionally dry, and this meant that Carl and the students were having to rely quite heavily on delay and reverb effects at the mixdown stage. Tom & Mike suggested that hardboard panels could be used to reintroduce some acoustic reflection when recording things like drums and acoustic guitars to make this less of a problem, and also recommended that Carl look at acquiring one of the recent high-quality convolution-based reverb plug-ins.

Speaking of reverb, Tom and I felt there was too much of it on the horn tracks, giving rather a long reverb tail which seemed rather out of keeping with the overall sound of the mix. The first thing to do was to take off all the processing and check the balance of the three mic signals. After a small bit of track rearrangement we managed to get the three channels up on adjacent faders, and it turned out that only a little re-balancing was required to get the horns to sound more 'authentic' — funk horn sounds are often pretty dry, so there was little need to add reverb. I also thought that the compression settings used were a bit harsh, and compromised the punchiness of the original dynamics. I felt that switching off the dynamics processing was an improvement, and that where the odd phrase poked a little too far out of the mix it would probably be best to sort this out with the VS2480's automation at mixdown.

Tom pointed out that the rhythm guitars sounded quite middly and were having to compete directly with the horns, muddling the overall mid-range. Carl and the students had already EQ'd the guitars quite severely — low-end shelving and high-frequency boost — to get them to cut through more, but there was more that needed to be done. The EQ processor's high-pass filter proved a better tool for the job, allowing us to progressively remove low end until the mid-range cleared up satisfactorily.

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Carl's Comments

"I really appreciate the guys from SOS coming down to the college to help us out. We all learnt an awful lot about the capabilities of MIDI, and also received some helpful tips regarding our studio setup. Since the visit we have progressed on from just using the VS2480 as a mixer surface to control volumes and panning, and we're now using the faders to adjust individual channel settings (EQ settings, effect sends, dynamics controls, and so on) and the parameters with *Hypersonic* and *Reason*. We're also going to begin to investigate using the VS2480 to control the parameters in our Korg Triton as well. Is there no end to this unit's capability? I hope not, as it gives us something to do in the classroom... Thanks again!"



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Homeward Bound

The flagship digital multitrackers are complex beasts which present a serious learning curve to the home studio owner, and computer sequencers are even more complex, trying to be all things to all people. So it's hardly surprising that getting a multitracker to work with a software sequencer can be a headache. Most apparent hardware/software faults in the home studio are the result of incorrect setup — indeed, it was an erroneous MIDI port setting that threw a

spanner in the works when we were trying to synchronise Carl's PC sequencer to the VS2480. However, the good news is that you can solve many studio configuration problems just by approaching them methodically, and a good technique here is to eliminate individual elements of a malfunctioning system from consideration until you can home in on the exact culprit, just as we did by testing the *Cubase SX* and VS2480 MIDI functions with the QY700 this month. Even the most complicated studio conundrums can usually be resolved if you can whittle them down to their root cause. **SOS**

Published in SOS May 2005

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Studio SOS

Above & Beyond/Anjunbeats

Published in SOS April 2005

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People : Studio SOS

The team are back in London this month to help some high-profile remixers sort out the monitoring problems in their new studio.

Paul White

This month saw us back in London at the premises of a team of dance-music composers and remixers who go by the name of Above & Beyond, and who own and operate the Anjunbeats record label. This group comprises Jono Grant, Tony McGuinness, and Paavo Siljamaki, and they've already enjoyed a healthy amount of commercial success, with regular appearances in DJ Mag's top 100 and remixes for Madonna, Dido, Ferry Corsten, Delerium, and Zoe Johnston. They are Radio One regulars and regularly DJ at serious clubs such as Cream, Crasher, and Godskitchen. The latest news is that their nomination for the BBC Radio One Essential Mix has just been announced the outright winner, which is probably why the guys brought in two packets of Hob Nobs rather than just the one!



With the speakers mounted on their stands, reflections from the computer monitor and gear racks were interfering with the sound. By moving the speakers forward onto the desk, Hugh reduced this problem, and he also placed bricks under them to put the drivers at a better height for seated listening — these bricks also reduced the resonance of the table top on which the speakers were standing. For long-term use, Paul suggested routing holes into the table top and fitting the original speaker stands through the holes.

The Above & Beyond studio had just moved into North London premises previously occupied by another small studio, and it comprised a single room with a vocal booth built into one of the rear corners, plus adjacent office space. Although the previous owners had applied a generous amount of acoustic treatment, this seemed to comprise a fairly thin layer of Rockwool covered by

fabric and it wasn't in the best place to be effective. For example, the side and front walls were damped only above the head height of a seated engineer; the rear wall and ceiling were untreated; and the vocal booth had the same thin treatment on the rear wall, ceiling, and one side wall, with a reflective door, window, and side wall left untreated. Apparently the original owner's monitoring system had been soffit-mounted Tannoys (inset into the wall), and the vacated soffits had since been completely filled with Rockwool, which meant they now functioned reasonably well as bass traps, albeit without intention in that direction.

For monitoring, the guys had a pair of Mackie HR824s augmented by a Mackie subwoofer, while the main recording system was *Logic Pro v7* running on a Mac G5 and viewed via a splendid Apple 23-inch LCD monitor. However, it turned out that the team have very strong ideas about audio quality and prefer the sound of analogue mixing to digital mixing, so they bring 24 tracks of audio out of *Logic* via three RME ADI8 units hooked up to an RME Hammerfall soundcard. These feed into a Soundcraft Ghost analogue mixer. They were also experimenting with some high-end preamps and had already come to the conclusion that their impressive collection of analogue synths sounded noticeably better when fed into the audio interfaces via a good preamp rather than being plugged in directly. They were experimenting with a loaned Neve VR preamp while we were there.

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Preliminary Listening

We were greeted by Jono, who explained that their monitoring sounded wrong, with an inaccurate low end and poor stereo imaging, and that the vocal booth was unusably boxy. After the obligatory coffee and biscuits, we first turned our attention to the Mackie HR824s, which were mounted on good-quality stands behind the custom-built furniture that Tony had designed, but this turned out to be less than ideal, as the two rack units and the large LCD screen were in front of the monitors where they could reflect and diffract the sound. It also placed the monitors at a fairly shallow angle to the desk surface, inviting surface reflections from the table top. The subwoofer was placed on the front wall off to one side in a somewhat inaccessible spot behind one of the built-in floor rack pods, though it eventually turned out to be close to the ideal position as far as the sound was concerned. A footswitch had been rigged up to bypass the subwoofer, which made setting up a lot easier for us.



In order to test the bass response of the monitoring system, a semitone scale sequence was created in *Logic* triggering a synth sound with the filter set low to remove upper harmonics. However, this proved unreliable in practice, as the levels of the upper harmonics varied unpredictably, so the *EXS24* software sampler was used instead — its default sound is a straight sine wave, which was ideal for the purpose.

Listening tests soon showed the bass end to be rather too hot, and the stereo imaging was also extremely poor. As the studio was said to be producing bass light mixes, the over-hot bass end didn't come as a surprise. We decided that the best course of action would be to drag the subwoofer into a slightly more accessible position so that we could reach the controls. We also decided to move the speakers off their stands and mount them on the desk surface, around a foot forward of their current positions. This would move them in front of the LCD monitor and the two side rack pods, which would reduce the diffraction and reflection problems significantly. However, the table top was far too low for them, so Sachi, the studio manager, managed to rustle up a few fairly clean bricks from somewhere to help add mass and damp any table-top resonances, while also raising the speakers sufficiently to place the tweeters near ear height and angled towards the listening position. During this relocation, we checked the HR824's rear-panel switch settings and discovered that the HF gain had been turned down by 2dB because Jono felt the speakers were slightly over-bright. This was fair enough, but we also found that one of the speakers was set to the correct 'half space' mode for use close to a wall, but the other was set to 'full space', which would have thrown the low end out of balance, at least when the speakers were used without the subwoofer.

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Subwoofer Settings

We settled on starting with the speakers set completely flat and with the room-position switches set to 'half space'. Once the subwoofer was dragged into a position where we could get at the controls, we tested it with a two-octave chromatic sequence of bass notes played via a *Logic* synth with the filter closed down to give a sine-like tone. However, the varying level of additional harmonics from this sound source became a distraction, so we switched to using the *EXS24* sampler, as its default waveform is a pure sine wave. The subwoofer crossover had been set to its lowest setting of 55Hz, but we reasoned we'd get more headroom from the system if we moved it up to 80Hz, as less low end would be fed to the main speakers. (The Mackie subwoofer crossover feeds a high-pass signal to the main speakers, and when the footswitch bypasses the subwoofer, these filters are taken out of circuit.) As expected, the subjective level of the scale got progressively higher as the pitch increased, but we noticed that the level climbed, dipped, and then climbed again, so we reasoned that something was happening around the crossover point.



An curious dip in the bass response led Hugh to suspect phase problems between the subwoofer and the monitors. A quick trip under the desk to flip the phase-inversion switch sorted out the problem, but further careful listening and adjustment was required to get the subwoofer level right in relation to the monitors.

Hugh set the phase switch on the subwoofer to invert, which cured this dip, so it

was clearly the best setting for this particular subwoofer position. Because of the layout of the furniture and the cabling to the subwoofer, it wasn't practical to do the usual trick of hauling the subwoofer to the monitoring position, then crawling around the room to find the most even-sounding spot, but if we had, I've a feeling it wouldn't have been too far from where the subwoofer was anyway. It certainly sounded fairly even and well balanced.

Getting the subwoofer level right was trickier, especially as the guys had turned the HR824 input gains quite a long way down to compensate for the unusually high output level from the Ghost desk. After much messing around, we decided it would be best to simply turn down the monitor output from the desk and then start with the main and subwoofer speakers set to their default 'detent' positions. When switching back and forth between having a subwoofer and no subwoofer, this sounded just slightly bass heavy at the upper end of our test scale where we reckoned there should be little difference with or without a subwoofer, so we backed it off slightly until the subjective level was the same. This added more depth to the very low notes but didn't significantly affect the higher bass registers. Tony found a sine sweep test signal and this again seemed to confirm that we'd got the settings about right. To make the subwoofer easier to move on carpet, we suggested fitting it with self-adhesive PTFE glider pads that are sold in DIY shops as an alternative to castors. Hugh fitted these to his A100 Hammond organ and Leslie, which he can now move on carpet with just one hand, so they do work.



The stereo imaging was still quite messy even after the monitors had been moved to better positions, so acoustic foam was fitted to the ceiling and side walls to minimise interfering early reflections — the foam panels on the right-hand side covered a window, so they were fixed with velcro to make them removable when natural light was more important than monitoring accuracy.

Hugh suggested that, as the console monitor output control was now less than a quarter of the way up at normal listening levels, attenuator pads should be fitted on the desk outputs to reduce the signal by 10-20dB, so as to allow the desk control to be set to a more realistic position and also to avoid the monitors flying off their perches if someone turned the level up full! In-line balanced XLR attenuators are readily available from suppliers such as Canford Audio and Studiospares.

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Stereo Imaging Improvements

Now it was time to improve the imaging by applying a relatively small amount of four-inch acoustic foam, kindly provided by Auralex. In small rooms where an 'empirical' approach to acoustic treatment is appropriate, the most successful

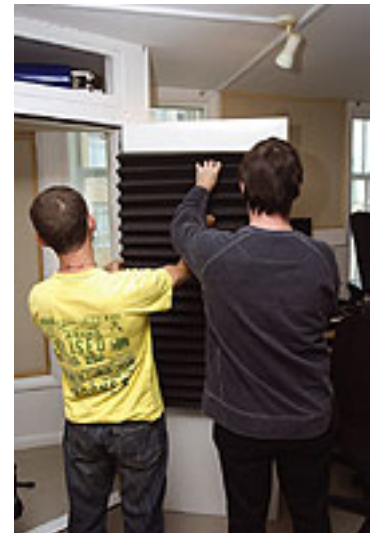
approach seems to be to locate absorbers on those surfaces that are currently reflecting sound from the monitors directly back to the listening seat. You can verify you have the right places by getting somebody to hold a mirror flat against the wall and then move it around until you can see the monitor from your mixing chair. This position determines the centre of where the absorber needs to be. Do this for both side walls and the ceiling.



A few offcuts of acoustic foam were pressed into service to reduce reflections from the wall behind the monitors.

We found that, in the case of the side walls, the foam absorbers needed to be placed below where the previous occupants had installed their shallow Rockwool absorber panels. On the left-hand side the ideal place happened to span a window, so Velcro was used to make that particular panel detachable. The permanent panel was fixed easily to the wall with spray adhesive, but the self-adhesive Velcro strips we used on the other panel were initially less than successful, as they kept peeling away from the foam. To solve this, we tried spraying a thin coating of adhesive onto the foam, waiting for it to go tacky, then sticking the Velcro to that — it worked a treat. All we needed in the end were three 2 x 4-foot panels, one on each side wall and one on the ceiling. One little tip here: if you get spray adhesive or mastic gun contact adhesive on the exposed side of the foam, you can usually get it off quite cleanly while it is still wet by dabbing at it with the sticky side of gaffer tape — but don't let it dry first.

Listening tests confirmed that the imaging had improved quite dramatically after moving the monitors and adding the new foam absorbers to the sides and ceiling — the last making a surprising amount of difference. The rear wall was partly obscured by a settee — always a good feature in a studio (both ergonomically and acoustically), but there was also a small window onto the adjoining office that acted as an unwelcome reflector. As the window wasn't really needed, Jono said that hanging a curtain over it would be a simple option. To take the edge off the remaining bare wall, I applied a 2 x 4-foot sheet of self-adhesive, one-inch dimpled foam donated by Sonic 8 — this is too thin to be used as the sole treatment in a room, as it is only effective at high frequencies, but when it is used to augment existing treatment, it works fine and sticks well to flat surfaces. A final touch was to cut a seven-inch strip of the self-adhesive foam and use this to cover a flat wooden strip directly behind the monitors.



A panel of thick acoustic foam reduced the boxiness of the vocal booth nicely, but there was still no life in the sound at high frequencies.

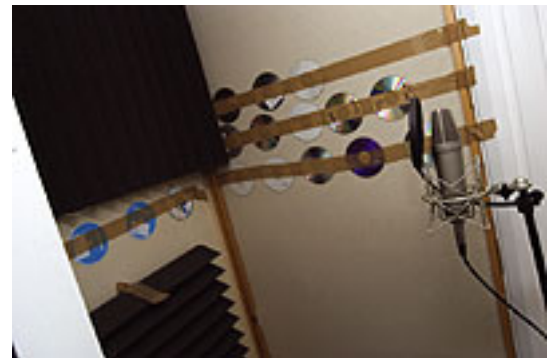
We didn't feel that additional bass trapping would be necessary, as the filled soffits, the doors, and the double-glazed windows already acted as bass traps, though there was a void above the vocal booth that could be filled with Rockwool if further bass trapping were to be needed. The main purpose of bass trapping is to ensure an even bass response, rather than simply to soak up bass, and we'd already achieved that as well as we could hope for in a room of this size.

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Vocal Booth Treatment

With the studio sorted out, it was time to take a look at the vocal booth, and we tested this by setting up a mic and getting Tony to do a spoken monologue while we listened in the control room. Sure enough it sounded dull and boxy. We tried again with Jono speaking, just in case Tony had a dull and boxy voice (which he didn't!), then set about looking for the cause and a possible solution.

Most bad-sounding vocal booths are made that way through too much absorption in the upper mid-range and high frequencies, with inadequate absorption at the lower mid-range and bass end. This can be due to using too great an area of treatment and/or using treatment that is too thin. In this case it was clearly the latter, as the inch or so of Rockwool that had been used would become progressively less effective as the frequency fell below 800Hz or so. My own view was that we should use thicker absorbers behind the singer (on the rear wall) and also try to introduce a little high-frequency scattering or diffusion. We could also use patches of self-adhesive foam just to take the edge off the areas of completely untreated surface within the booth.



By taping old demo CDs to the walls of the vocal booth at around head height, enough high-frequency reflection was reinstated to provide a nicely balanced overall sound.

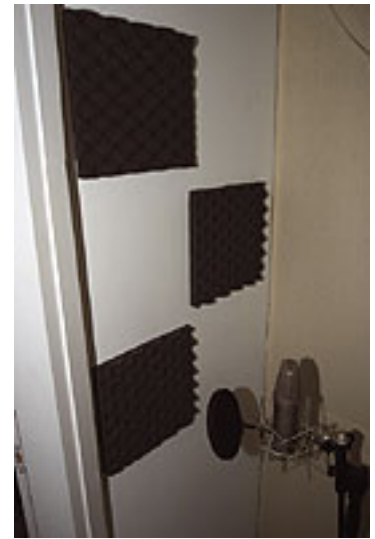
The upper part of the rear wall included a cavity for a light (which wasn't really necessary), and it seemed that this was contributing to the boxy feel. To test the theory, I wedged a piece of four-inch Auralex foam across the top of the rear wall and stood another piece on end at the bottom of the rear wall. A further 2 x 4-foot panel was glued to the inside of the hard door. This certainly dried up some of the 'honk', but the top end was still lifeless. We needed to improvise some high-end scattering, but what could we use? What does a small record company have lots of that are small and sonically reflective at high frequencies? Demo CDs, or course!

Just to test the theory, we taped a couple of rows of CDs to the rear wall and to the padded left side wall at about head height. It really made a difference, adding just the right amount of brightness and presence. The spoken monologues now

sounded much more natural, with none of that 'broom cupboard' sound that had beset the booth originally, so we declared the experiment a definite low-budget success. The guys said they'd do the job more permanently and more neatly using spray adhesive. They quite liked the look — using discs with nice label artwork would look pretty good and could be made into a nice feature. There was also room to put a few CDs on the ceiling if necessary.

We noticed that the door of the booth didn't seal too well and so suggested fitting a compression door latch that would pull the door up against the seal to make it a better fit. There was also no permanent connection panel in the booth (the mic cable had to be run through a gap at the bottom of the door) so this was also added to the 'to do' list.

The temporary speaker relocation on bricks had worked well, but was not a very satisfactory long-term solution from an aesthetic point of view. So we suggested cutting a couple of large circular holes into the desk using a router, directly below where we currently had the monitors set up. They could then use the original speaker stands, projecting up through these holes (the top and bottom plates can be easily removed to facilitate this) to support the speakers in a more aesthetically pleasing way, while also isolating them completely from the desk. Once again we'd shown that you can make a big difference to how well a studio room performs for relatively little cost and with minimal disruption. Hopefully the guys will now get even better mixes from their new studio than from the old one. **SOS**



A few thinner pieces of foam were placed on the wall behind the singing position to further reduce boxiness.

Published in SOS April 2005

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Studio SOS

Tom Lindsey

Published in SOS March 2005

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People : Studio SOS

The SOS team travel to an attic in Birmingham to help a drummer improve his recordings.

Paul White & Hugh Robjohns

Tom Lindsey is a student at the Birmingham Conservatoire as well as being a drummer with a home studio. Tom's studio is based around *Pro Tools LE* and a Digidesign 002 interface/controller powered by a 3.2GHz Windows laptop computer. Although he has little in the way of additional equipment, he does have a nice pair of Mackie HR624 monitors and some carefully chosen microphones. His studio is located above a garage, which has resulted in a studio space with sloping walls and a narrow flat ceiling section just below the apex of the roof. At the time of our arrival there was no acoustic treatment in the room other than a settee, but Tom had bought three large Auralex foam panels in anticipation of our arrival.



The foam isolation pads underneath Tom's Mackie HR624 monitors had been positioned such that the monitors were pointing downwards, away from the monitoring position. To remedy this, Paul suggested turning the foam pads around so that the speakers pointed slightly upwards towards a listener seated at the desk. The shelf above the speakers was also causing high-frequency reflections, so Paul moved the monitors a few inches forward to minimise this problem.

Although we have had a lot of SOS visits recently that have centred on studio acoustics, Tom's problem was not monitoring, but rather getting a good drum sound in his relatively confined environment. The solution involved a mixture of acoustic treatment, microphone positioning, and signal processing. However, we did spend a few minutes optimising his monitoring before we got stuck into the main task. We also got stuck into his extra large packet of chocolate Hob Nobs before looking at his monitoring!

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Getting To Grip With Gripps

Tom's studio setup has him working across the width of the room, which places the side-to-side reflective surfaces well away from him, but the wall behind him is relatively close. The settee behind his mixing seat absorbs some sound, but using the 'mirror' principle, it was evident that there was a strong reflection path where the sound from the monitors bounced off the sloping ceiling and back to his listening position. To tackle this reflection, we decided to fix one of the Auralex panels to the sloping ceiling behind him, but as he hadn't bought any adhesive with the Auralex foam we set off to the local Focus DIY shop to seek an alternative. A helpful young man directed us to the flooring section and showed us a spray adhesive called Gripps Spray, and though this was specifically designed to fix vinyl tiles he said it should be fine on foam. As it turned out, it was only partially successful — even when we applied it to both the walls and the back of the foam, it had a tendency to come unstuck after lulling us into a false sense of security for a few minutes! It would be best to use the proper Auralex adhesive or something like Liquid Nails applied using a mastic gun. The Gripps Spray also managed to fill the air with finely atomised glue, which was very unpleasant, not to say somewhat unnerving!

Tom had his monitors stood on high-density Auralex pads to help isolate the vibrations from the table they were resting on. However, as the speakers were standing on the desk rather than on tall stands, they were below the optimal listening height, and the angled pads made them point down towards the table top. To remedy this, we simply turned the pads around under the speakers so that they were facing slightly upwards, with the tweeters pointing towards the ears of a seated listener. We also moved the speakers forward by a few inches to avoid high-frequency reflections being created from an overhanging shelf. The Mackie monitors were set with their EQ controls flat and the environment-compensation switch in the 'half space' position. With these positional tweaks the imaging improved by a worthwhile amount.

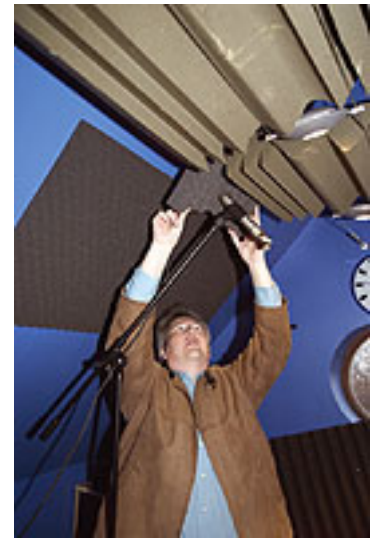
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Drum Recording

The drum kit, on the other hand, was to prove more of a challenge. Tom's nicely tuned Pearl drum kit was set up in one corner of the room facing across the width of the studio. An AKG D112 was used to mic the kick drum, a Shure SM57 the snare, and a Rode NT4 stereo mic was used to capture the overheads. Tom didn't have any decent tom mics, but said that he didn't use the toms much anyway, so there was no need to mic them separately. We listened to some of Tom's existing recordings and agreed with him that the room sound was rather too intrusive, exactly as he'd described when he called us. Given the number of flat, reflective surfaces surrounding the drum kit, that wasn't surprising. The recorded kit sound was also a little dull, especially when compared to the live

sound.

We reasoned that our first priority was to cut down the amount of reflected sound reaching the overhead stereo mic, so we set out to suspend another of the Auralex panels from the ceiling. Running the length of the room, just a few inches below the centre of the ceiling, were a pair of wires from which the low-voltage lighting system was suspended, and it was a simple job to place a slab of foam on top of the wires where it would be suspended around four inches from the ceiling. The NT4 stereo mic was then set up towards the back of this piece of foam so that the foam would intercept reflections from the nearby ceiling.



In order to dry up the overhead sound, Hugh suspended some acoustic foam from the ceiling and attached more foam to the back of the overhead mic.

We noticed that Tom didn't have shockmounts for any of his mics, but he had bought one of the Auralex kits that comprise foam supports for mic-stand legs, as well as small foam baffles that can be clipped onto the backs of microphones to cut down on off-axis sensitivity at mid and high frequencies. We used the mic stand supports to provide some isolation for the overhead and snare mics, and then used the spare speaker-pad wedge sections (normally used for fine angle adjustments) to float the kick-drum mic stand. As usual, the kick-drum mic was set up inside the drum shell and offset a little to one side of the beater about halfway into the drum.

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Overhead Miking Tweaks

A test recording made with this simple setup gave a good basic kick sound, although slightly lacking in 'click', a somewhat dull snare sound, and an overhead sound that was drier than before, but still not adequately tight sounding. If drums are being recorded in a great-sounding room, then the sound of the room contributes to the final result in a very positive way, but where the room sounds boxy or coloured, it's safer to exclude as much of it as possible, even if it means adding artificial reverb later.

To dry up the sound further, we glued another large Auralex panel to the sloping wall above the drum kit, as sound coming off the snare and toms was reflecting off the angled ceiling and heading off in the direction of the overhead mic. We'd also brought along a couple of large XXL AC01 self-adhesive foam panels contributed by Sonic 8, the UK distributors, so we stuck one of these to the sloping ceiling opposite the drum kit, and the other to the wall directly behind the kit.

We had a couple of 18-inch-square, four-inch-thick foam panels left over from a

previous job, and Hugh had the idea of putting these on top of the lighting wires to release the larger Auralex panel from this role, relocating that panel to the end wall of the room adjacent to the drum kit, helping to kill reflections off that wall as well.

With these panels in place, we repeated the test recording and found that the sound was much drier and noticeably less coloured, though the snare sound was still dull and the overheads lacked a little sparkle. Given that we'd done about as much as we could with acoustic treatment, we resorted to fine-tuning the mic positions. As they were available, we also placed the little Auralex foam baffles behind the overhead and snare mics to see if we could dry up the sound a little more.



Foam was fitted around the snare mic to reduce spill, and more foam was used to isolate some of the mic stands from mechanical vibrations, as none of the mics used had proper shockmounts.

The snare mic was set up in its usual position, about two inches above the drum head, just in from the rim and angled towards the centre of the drum. But as this was sounding dull, we raised it to four inches to reduce the proximity effect, which was contributing an unhelpful bass boost. This repositioning helped, but it still needed EQ to get it as snappy as we wanted it. As an alternative approach we also tried miking the snare side of the drum (and tried both settings of the channel phase-reverse switch), which gave a very snappy, but at the same time thinner, sound. When combined with the overhead mic, the combination added bite to the snare but made it sound somewhat brittle, and even though the sound was better with the mic output phase inverted, we decided that the top-miked position was still the best approach overall.

There was little we could do with the overhead stereo mic other than angle it down more so that it was aimed at the cymbals. Ideally it should have been higher and there should have been plenty of space between it and the ceiling, but in this room it simply wasn't possible. The foam would have the effect of raising the ceiling to a degree, but it was time to resort to processing. The most dramatic improvement was brought about by EQ'ing the two overhead mic tracks. A broad 5dB boost at 11kHz did wonders for the sparkle of the cymbals, while cutting below 64Hz by 6.5dB using Digidesign's low shelving filter thinned out the low end and clarified the sound. Adding the kick mic back in restored the necessary weight, but there was so much snare sound in the overheads that the close mic made less of a difference to the overall tonality than we would have hoped for.

To get a more contemporary bite from the kick sound, we boosted it by 9.5dB at

4.5kHz and used the Digidesign gate (set to its fastest attack) to clean up any ring. These EQ values may seem extreme, but the Digidesign EQ seems to be one of those plug-ins that has to be applied quite generously to make any significant difference.

To brighten up the close-miked snare sound, we added a whopping 10dB of boost at 6kHz, and though this sounded a little extreme in isolation, it was far more natural sounding when the overheads were brought back in. A

gate was used to clean up the ring from the snare drum, and to put a little life back into the snare sound we added some Digidesign *D-Verb* reverb (medium plate). We had to add quite a lot to make the reverb audible once the overhead and snare mics were mixed, but it definitely opened up the sound and made it more believable. Ideally I'd have liked to try some more sophisticated ambience programs, but the *D-Verb* didn't seem to have anything particularly well suited.



As an attempt to dry up the sound further, Paul fixed more foam to the wall behind the drums.


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A Good Compromise

Although the drum sound we ended up with was still something of a compromise, it actually sounded pretty good and was a vast improvement on the sound recorded in the untreated room. Tom still wasn't entirely happy with the snare sound, but conceded that what we were now getting sounded pretty much the way the drum really sounded. He'd recently experimented with different head types and felt that he might have to do further experimentation to get it sounding the way he wanted it. The simple foam absorbers killed most of the undesirable room coloration, while the EQ we applied to the overheads did the most to lift out the cymbals, add sparkle, and reduce low mid-range mush. Even the toms sounded pretty good on the final recording, even though they had no separate mics.

Tom happened to mention that he usually records vocals at the drum end of the room and that he felt the sound would be better now that there was more acoustic treatment on the walls. He had been intending to face the singer in towards the acoustic foam, but we explained that when using cardioid mics, it's invariably best to position the singer with the absorbent material behind them so as to cut down on reflections getting back into the mic. Furthermore, hanging an extra duvet behind and to the sides of the singer is worthwhile in smaller rooms, as there's nothing worse than spending good money on a decent microphone and then ending up with a boxy vocal sound. In this particular room, it should be easy to rig up a sturdy curtain rail from which to hang a duvet.

Our final observation was that, as the room was currently unheated when not in

use, some form of low-level background heating might be useful to prevent the capacitor mics from getting too cold, as they are quite susceptible to condensation before they get up to room temperature. The other option would be to keep them in the house and bring them into the studio only when required. 

Published in SOS March 2005

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SOUND ON SOUND

The World's Best Music Recording Magazine

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Studio SOS

Aniff Akinola

Published in SOS February 2005

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People : Studio SOS

Acoustic problems and mains hum were making Aniff's studio difficult to use, so the SOS team stepped in to help.

Paul White

Aniff Akinola has been working in music for around 20 years and has enjoyed a certain amount of commercial success with his productions, including collaborations with artists such as the late Kirsty McColl, Steve Hillage, Ian Brown, The Beloved, A Guy Called Gerald, Urban Cookie and Backyard Dog. During that time he's run two independent record labels, been signed as an artist to four major labels, and been involved in the creation of three top-20 hits. He's currently signed as a writer to BMG Publishing. As if this weren't enough, Aniff is currently working part time at Manchester's Sound Control music store.

He called us in because he wasn't happy with the monitoring environment in his studio, located on the top floor of his Old Trafford home. Prior to our arrival, he'd bought a pair of Genelec 8040As and a monstrous Genelec 7070 subwoofer — along with a pack of Primacoustic room-treatment foam. The subwoofer was serious overkill for



Here you can see Aniff's setup as Paul & Hugh found it on their arrival, and this configuration was causing several problems. For a start, the Genelec monitors were too far apart, compromising the stereo imaging at the monitoring position. The central monitoring position wasn't helping the bass response, and the subwoofer (to the left of the desk in the lower picture) also needed a little repositioning.

the size of room, but it was a bargain secondhand buy, and Genelec equipment is always easy to set up in almost any combination. Our job was to help him optimise his monitoring environment, sort out his subwoofer placement and balance, and fit the foam where it would do most good.

When we arrived, Aniff had set up his Genelecs on stands behind a pair of Quiklok studio desks on which he'd placed his Yamaha NS10s plus a pair of flat-screen monitors for use with his Apple *Logic*/G5 studio setup. His G5 is fitted with both TC Powercore and Universal Audio UAD1 DSP expanders and he uses a Rode Classic 2 as his main recording mic. To keep the Genelecs clear of the NS10s, he'd had to set them very wide apart, which was less than ideal from a stereo-imaging point of view, and he'd also reported some hum problems with his NS10s, which were driven from an old Quad 405 MkII amplifier. He'd parked the subwoofer almost exactly halfway along the left wall of the room, and the desks were a couple of feet from the front wall to make space for a large MDF box that housed his Mac G5 and a PC. This arrangement resulted in the listening position being almost exactly in the centre of the room.



Monitoring tests proved frustrating working directly from the outputs of Aniff's MOTU 828 MkII audio interface, so Hugh quickly plugged up Aniff's new Samson C*Control monitoring controller to make switching and level control easier. The next step was to make sure that the Genelec monitors were suitably set up, and this involved experimenting with the subwoofer positioning, level, and phase.

The wiring behind the desk resembled the pit of snakes from an Indiana Jones film, and a rather lost-looking Samson C*Control was perched hopefully on the desk with no cables yet attached to it. A MOTU 828 MkII handled the audio interfacing, and a Focusrite ISA430 front end was connected to the S/PDIF input of the MOTU when required. One of the few hardware synths to survive Aniff's transition to software was a Roland JV2080 fitted with a number of expansion cards, though Aniff also has an Akai MPC4000 that he likes very much.

Preliminary Listening & Subwoofer Placement

As usual, our first job (after attending to the tea and chocolate biscuits) was to listen to the monitoring system exactly as it was. Initially, this was done using the MOTU 828 MkII's volume control, but this proved frustrating to control, and the resolution fell off at low volume settings. So Hugh quickly set about patching in the Samson C*Control instead, with the MOTU monitoring outputs set at full level to achieve the best D-A resolution, and with Aniff's CD player hooked up to one of the two-track inputs so that he could play back CDs without re-patching.



Following a quick trip to B&Q, Paul constructed an acoustic panel using a sheet of MDF, some adhesive, and elements of a Primacoustic foam room-treatment kit.

As expected, the stereo imaging was less than ideal, because of the wide speaker spacing, the very reflective side walls, and the relatively large amount of clutter in front of and between the monitors on the desk — in fact it was so bad that we could detect very little difference when we pressed the C*Control's Mono button! The bass end also seemed rather too loud and lumpy. We reckoned that the imaging would be improved by moving the speakers and by putting up the Primacoustic foam Aniff had bought, so Hugh set about adjusting the subwoofer while Aniff and I made a trip to the local B&Q to buy some fixings for the MDF panels on which we'd decided to stick the foam. As there was a fair bit of gluing and lifting to do, Aniff conscripted his friend Ben from Manchester's branch of Sound Control to help out.

If the subwoofer isn't in the right position in the room and relative to the listener, the level of differently pitched bass notes can vary enormously, so Hugh and Ben used the old trick of lifting the sub into the place where the engineer's chair normally goes, then crawling around on the floor alongside the walls listening for the spots where the bass sounded most even. There were a couple of places which seemed quite good, and one turned out to be not far from where the sub had been located in the first place — it just needed moving forward by a foot or

so, which got it away from the 'dead centre of the wall' position, which is usually less than optimal.

Having located the ideal place, the subwoofer was dragged back over and then the DIP switches that control the relative phase were checked to get the best crossover response with the 8040s. Finally a listening test was performed to establish the correct bass balance. It was necessary to drop the sub level by around 2dB from its original setting to get a subjectively sensible bass balance, though this wasn't easy to judge, as Aniff's collection of predominantly drum & bass music seemed to vary rather a lot when it came to bass content. Hugh managed to find some more 'known quantity' pop albums and we eventually agreed on a setting that was a happy medium.



Aniff held a vacuum cleaner to catch the dust while Paul drilled holes for mounting the acoustic-foam panel he'd just put together. Once the hole was drilled, it was fitted with a wall plug to accept the mounting screw. A picture bracket was then screwed to the back of the panel so that it could be fitted to the wall screw's head.

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Killing Undesirable Sonic Reflections

Aniff decided to fix the Primacoustic foam to MDF panels so that they could be moved around if necessary and when redecorating the room. As luck would have it, we were able to make up a pattern of foam blocks that exactly fitted the 2 x 4-foot MDF panels he had bought. Using a panel adhesive and a mastic gun, we fixed up four identical panels to be used in pairs on the side walls to tighten up the imaging and kill flutter echoes. A further panel was used directly behind the monitors as a horizontal absorber, with four thick foam blocks overhanging the panel at either end. The remaining foam blocks were fixed to a sheet of MDF and placed on the rear wall, though some additional rear-wall absorption would be beneficial. Aniff said that he planned to get a settee to put at the back of the room, so that should also help kill reflections. In addition, we discussed the idea of hanging a couple of single duvets across the angled part of the ceiling coming down to the back wall, which Aniff said he'd try to do later on.

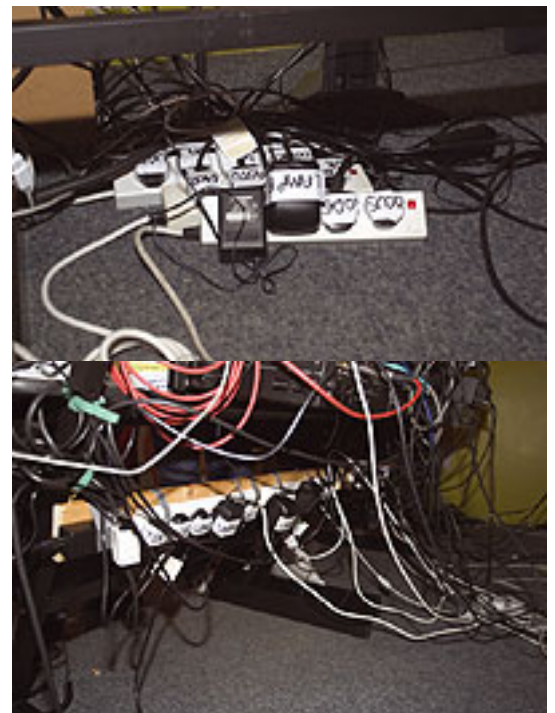


In an attempt to even out the room's bass response, some foam bass traps were also installed directly into the room's corners using adhesive.

We fixed the panels to walls using brass picture-hanging plates screwed to the backs of the panels. These simply slotted over screws in the walls that were fixed using plastic wall plugs in the traditional way. The two triangular foam bass traps that were included in the Primacoustic kit were glued between the front wall and the ceiling. Fortunately, bass trapping wasn't a serious issue in this room, as there was a panelled chimney breast in the front right corner, a door enclosing the hot water system in the front left corner, a wooden entrance door (back left) and a window (back right), all of which soaked up bass quite effectively and in a well-balanced way.

Once we had the panels in place, we experimented with the speaker positioning. Although the side panels had certainly improved the stereo imaging a little, it was still way below that which we expected from these Genelec speakers — and the NS10s sitting on the desk shelf produced a far better stereo image. Although it is generally better to place speakers on dedicated stands, rather than on the desk, in this case we found that with the Genelecs on stands behind the desk the clutter on the desk shelves (NS10s and LCD monitors) wrecked the imaging completely.

So, we eventually ended up with the NS10s directly to the sides of the two computer screens, then the Genelecs placed on the top shelves next to



Aniff's nest of adaptors and mains plugs were cluttering up the floor underneath the

these, so the stands were no longer required. Fortunately, the Isopod stands that are supplied with the new Genelec speakers are very good at helping to isolate vibrations to and from the desk, and the stereo imaging was now perfectly good. Aniff was worried that the NS10s might not sound right, as they were now standing vertically, but this is the way they are actually intended to be used. Mounting them on their sides is a convenience issue (to help lower their profile when mounted on a big desk meterbridge, obscuring the main soffit-mounted monitors) and narrows the sweet spot very noticeably.

main desk, so the SOS team improvised a long suspended mains connection strip using a piece of wood and several six-way plugboards. This cleared the cables off the floor, making access behind the desk (for re-patching or cleaning) much easier. Because the six-way plugboards were connected to a single surge-protected plugboard in a 'star' configuration, this setup also had the effect of reducing the level of mains hum in the system.

We have noticed before that there can be a bass suck-out near the centre of a square room and, because the two desks were a fair distance from the wall, Aniff's chair was dangerously close to this dead spot. In fact, this centre-room 'hole' became very noticeable as you stood up or sat down. The solution was to remove the computer box from behind the desk to allow us to move the two desks closer to the wall. This improved things considerably, and the monitoring balance, especially at the bass end, was far more acceptable in this new location.

Ideally, we should have used balanced cables between the 828 MkII and the C*Control, and also between the C*Control and the Genelecs. However, Aniff didn't have any spare balanced cables, so we had to make do with unbalanced cables between the MOTU and Samson boxes as a temporary solution, with a promise that he'd change them as soon as he could. The main speaker output from the C*Control used balanced cables to the Genelec subwoofer and from there back to the 8040s.

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Curing An NS10 Hum

Having sorted out the main monitoring, we turned our attention to the humming NS10s. There was, indeed, a constant low-level hum, and we first assumed this was a ground-loop problem, as the Quad amp's inputs are unbalanced. However, disconnecting the inputs didn't change the hum level or quality at all, and neither did shorting the tip and ring of the phono connectors — if shorting the inputs had killed the hum, then the hum may have been getting in at the inputs.



The finished setup, with the monitor controller plugged up, the desk and monitors repositioned, all the acoustic treatment in place, and the hoover safely back under the stairs...

This all pointed to a problem in the amp itself — possibly faulty power-supply smoothing capacitors — and would require the services of a good repair centre. However, there was also a second possibility. The amp had been modified at some stage to install an input level control, and it was possible that the wiring for this was picking up a stray hum field within the amplifier. Either way, Quad Electroacoustics maintain a very good repair service, so we suggested Aniff contact them to have the amp overhauled.

The speaker leads to the NS10s were also enormously long, which was pointless and could only reduce the sound quality, so Hugh quickly shortened the cables and we then hooked up the Quad amp inputs fed from the Monitor B output of the C*Control. The Samson has facilities to balance the levels of the two main pairs of speaker outputs, so that switching from the Genelecs to the NS10s maintained nominally the same level.

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Tidying Up The Wiring

While we were at it, we decided to have a go at tidying up Aniff's mains wiring. To get the mains connections off the floor, we bought some new six-way mains distribution blocks and taped these to narrow planks of wood that would fit between the upright legs at the rear of the Quicklok desks. Ordinary brass hooks were screwed into the ends of these, then they were simply hooked onto Velcro straps fastened around the rear desk legs, just above one of the horizontal supports to stop them sliding down. We did this on both desks, which gave us four sets of six-way distribution boards around 18 inches from the ground, and we also had enough Velcro straps left to help tidy up the cabling.

To minimise the risk of ground loops and to offer the maximum protection for the equipment, we ran all the distribution boards back to a central surge-protected distribution board plugged into a wall socket. This meant the whole setup was running from a single power point with the mains cabling in a 'star' configuration. Running everything from a single wall socket wasn't a problem, as the overall power consumption was fairly small. Aniff had labelled all his mains adaptors

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Aniff's Comments

"I still can't get over how different the room sounds just when I walk in, never mind when I'm playing music! I have now got the quilts for the rear ceiling and will put them up in the next couple of days. I've also put in a bypass switch for the sub for when I'm working late at night. Now it's time to go and practice my soldering!"



and plugs, so it didn't take long to get everything up and running again.

The result was a hum-free system (with the exception of the low-level hum on the NS10s from the suspect power amplifier) and, although the wiring was still far from being a work of art, at least it meant you could get a vacuum cleaner under the desks without encountering piles of cables! As we'd made a fair bit of mess sorting out his system, I tested this by vacuuming the room in readiness for our photos! By the way, please don't tell my wife that I know how to do this...

With the desks shuffled back, the C*Control repositioned centrally below the dual LCD screens, and the surplus cabling removed, the whole setup was better both ergonomically and sonically. The stereo imaging was actually very good, and the level of bass end seemed appropriate.

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Last-minute Soldering Tips

Our final task was to try to teach Aniff to solder using a propane-powered soldering iron we'd picked up at B&Q. I demonstrated the basics by replacing a stereo jack plug on his headphones, and I pointed out that the iron shouldn't be used to carry solder to the job. Instead, you heat the two parts to be joined separately and feed solder onto them to tin them. Then, the two tinned parts are brought together, heated again and more solder fed onto the joint as required, until the solder flows freely across the joint. At this point, the heat is removed and the joint held still until the solder has cooled. Aniff tried this on a couple of scraps of mains wire and seemed to be getting the hang of it by the time we were ready to go, though he had to be reminded to keep the heat on the joint and not dab at it with the iron. It only takes a few minutes of practice to learn to solder, and it is a hugely valuable skill, especially if you need to repair cables or make up custom connections.



Before he and Hugh headed for home, Paul used a propane-powered soldering iron he'd picked up at B&Q that morning to show Aniff how to go about basic soldering.

As the sun set over the sleepy suburbs of Manchester, Hugh punched the route home into his trusty GPS navigator, and we headed for that perpetual car park that is the M6! **SOS**

Published in SOS February 2005

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Studio SOS

The Loose Cannons

Published in SOS January 2005

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People : Studio SOS

The SOS team head to the 14th floor of a London tower block to help a pair of readers improved their drum sound.

Paul White

The Loose Cannons comprise Kaiser Saucy and Lord Fader, and the latter invited us to his lair (we thought it was going to be a hollowed out volcano with integral mono-rail and sharks with lasers!) to help sort out a few recording problems they'd been having. The pair have been DJs, promoters, remixers, writers, and producers for approaching four years now, but thanks to a brief and volatile marriage to a major label (recently divorced due to irreconcilable differences), the boys have at least been able to up their profile considerably in the last 18 months. Their new album *'Perbe* is due out in late spring 2005, the first fruits of which, including the single 'Raw', are doing the rounds now.



SOS reader Lord Fader re-plugs his Mackie mixer after Paul suggested that it would be better to feed his studio monitors from the dedicated control-room outputs, rather than from the main mix outputs.

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14th-floor Drumming

This particular situation was a bit of a first, as Lord Fader (his stage name, you understand, and just Fader to his friends!) has his home studio in a 14th-floor flat. Fortunately Hugh and I were able to take the elevator so as to avoid doing battle with the Death Stair... After acclimatising ourselves to the rarefied air of the

South London stratosphere, we were rewarded by the appearance of some excellent Sainsbury's triple-chocolate biscuits backed up by the now mandatory Hobnobs and a nice cup of hot tea. This was going to be a good day!

Fader's studio is based around a Mac G4 running Emagic *Logic* v6 using a MOTU 828 as an audio interface and a Korg Z1 as the main keyboard — all very familiar. He also had a Mackie 1604VLZ Pro mixer hooked up, but he was monitoring from the main outputs and re-patching just about everything for each new job, so we thought it might be best to rationalise that aspect of his system first.

His monitoring was via a pair of Alesis Monitor 1s, which were positioned rather asymmetrically in the room, so Hugh sensibly suggested that he move everything about a foot to the left to at least get the right-hand monitor away from the corner. DIY acoustic treatment and curtains gave the room a reasonably neutral sound, and the rest of the setup was pretty much OK.



The hole in the kick drum's front head was too small to get a microphone easily into the drum cavity, so Paul had Fader and Kaiser remove it.

While Kaiser assembled the drum kit, Fader and I set about re-knitting the mixer wiring so that the monitor amp was fed from the mixer's dedicated control-room outputs, and so that the main stereo out of the MOTU 828 fed into the two-track inputs on the Mackie. This configuration has the advantage that the control room section is isolated from the rest of the mixer if the two-track input is chosen as the only monitoring source, leaving the rest of the mixer (in this case two main outputs and four subgroup outs), free to feed into the MOTU's six line-input jacks. As this particular MOTU interface has two mic/line inputs and six line-only jack inputs, wiring the Mackie this way meant that all eight audio inputs could be fed with mic signals simultaneously if necessary (two direct and six via the Mackie).

Fader had made a note to ask us about hum and ground loops, as he was experiencing some problems with his setup, although this actually improved a little after the rewiring. However, all his jacks were unbalanced, and as both the Mackie mixer and the MOTU 828 have balanced line connections, using balanced jack cables should improve the situation; so balanced cables went on Fader's shopping list.

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Miking The Drums

Once the drum kit was set up facing the recording setup to give us good access to the mics, we decided to remove the front kick-drum head, as the small hole

that had been cut into it wasn't large enough to allow us much flexibility in mic positioning. Also, the damping (folded blankets) needed modifying to get a decent tone out of the drum. This done, Kaiser went on to apply a little damping to the upper and lower heads of the two toms using pieces of black gaffer tape. (Why is gaffer tape like The Force? Because it has a light side and a dark side and it holds the universe together!)

Knowing that we were going to be recording a drum kit, we took along a set of CAD drum mics that we'd had in for review — this seemed like an ideal opportunity to give them a practical workout. It turned out that the included drum shell clamps wouldn't fit the snare drum in this kit, but everything else went up smoothly and we used a standard boom stand for the snare mic. The clamping hardware worked perfectly on the toms, supporting the close mics a couple of inches above the edges of the drums facing inwards. While I fitted the tom mics, Hugh rigged the two electret overhead mics from the CAD kit as high as practical in the form of a spaced pair using two more standard boom stands, and finally rigged up the snare mic.

Initially, we tried resting the kick-drum mic on the pillow inside the drum, but the sound we obtained was very dull, so we used a second mic stand to support the kick mic about halfway inside the shell and slightly to one side of where the beater hit the head. The overhead mics were fed to the two mic inputs of the MOTU (and fed phantom power) while the snare, kick, and tom mics were patched into the Mackie mixer and then routed to four of the MOTU line inputs via the mixer's group buss outputs.

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Test Recordings

After setting the levels, we made a test recording directly to four mono tracks and one stereo track in *Logic*, with Kaiser playing the drums. We had no chance to monitor the sound prior to playback, as the drum kit was right next to us, but with no EQ or other processing, we achieved what was actually a very decent drum sound, with quite adequate separation. However, the separation on the kick-drum mic wasn't as good, and it sounded just a bit too leaden and soggy as well. Messing around with the mic positions helped a bit, but we eventually decided to switch to a Sennheiser MD421 instead. This produced a better balance of weight and click. Fader said that when he'd tried recording the kit, he'd had trouble getting any separation, and that all the mics had picked up more or less the same thing, so my guess is that he didn't have his close mics close enough.

After checking all was well with the CAD mics, and having proven that it was certainly possible to get a good drum sound in the room, we re-rigged the kit with the mics that the guys had available in the studio, to try to get similarly good results. We ended up using an SM57 on the snare, the MD421 on the floor tom and something that looked like an old AKG D12 on the kick. Very much a case of *The Usual Suspects*! The overheads were initially an old AKG C451/CK1 and a newer AKG BlueLine SE300 with CK91 cardioid capsule. However, the C451

seemed to have stopped working, so we used the SE300/CK91 together with an Audio-Technica AT4033 large-diaphragm mic instead.

These are obviously quite different microphones, but once the levels were matched, the subjective end result was actually rather nice. Capacitor mics are always preferred in this role, as their extended high-frequency response captures the transients and cymbals more effectively than dynamic mics. The shockmount for the AT4033 had seen better days — the upper elastic support band had broken, and the cradle elastics had stretched. Fader had boded together a replacement system using several large elastic bands which seemed to work well enough, even if it wasn't too pretty. He had also attached a homemade pop shield that seemed to work well for vocals too.



Initially the kick microphone was laid on a cushion in the drum, but this gave too dull a sound, so it was then mounted close to the batter head on a stand.

If anything, the hi-hat element of the drum sound could have been a little louder, but they didn't have a spare mic to record this separately, so it was picked up mainly in the overheads. Repositioning the snare mic may have helped, along with some judicious equalisation, to emphasise the hi-hat more, but for real control a separate hi-hat mic would be the best solution. Any small capacitor mic should be OK for hi-hat recording provided that it is positioned just above or just below the point where the two cymbals meet, otherwise the rush of air as the hi-hats close could upset the mic's performance and compromise the sound.

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Improving The Kick-drum Sound With Processing

The drum sound we achieved using their assortment of mics was a little tonally different to that achieved with the CAD mic kit, especially in the overheads, but still pretty good. However, we felt the kick needed more definition to fit in with their urban style of music, so after checking the mic position again and deciding we were getting about the best sound we were going to get, I tried a few *Logic* plug-ins to see what could be achieved.

I have to admit that I was feeling quite nervous at this point, as we'd been playing the drum kit on and off for over an hour and I half expected to see a crowd of locals with torches and pitchforks coming to get us. After all, this was a flat with neighbours above, below, and to either side! Fortunately, Fader had already

warned his neighbours of our visit and it seemed they were either incredibly tolerant or out at work during our recording session.

The starting point for processing was to use the channel EQ, where around 6dB of boost at 76Hz (the kick drum's fundamental) combined with a 5.1kHz boost of 8dB (to bring out the click) started to work quite nicely. A little mid-range cut at 280Hz to tame any boxiness also improved things, but it is important when doing this to listen to the kick sound both soloed and in context with the rest of the kit, as the overheads make a very significant contribution to the sound. The other trick I tried was to use Logic's *Exciter* plug-in to bring out the kick mic click, as this can synthesise high-end harmonics based on what's going on lower down the spectrum. Again this turned out to be quite successful. The same plug-in has also proven useful in the past for rescuing dull snare tracks.



The toms needed a little damping with gaffer tape before they were close-miked by Hugh.

As a final kick-drum experiment, I opened Logic's *SubBass* plug-in, which creates frequencies an octave below those selected for analysis. You can think of this as being like the *Exciter* plug-in in reverse — instead of synthesising harmonics above those that are actually there, it synthesises sub-harmonics below those present in the program material. There are two sets of controls, so that you can generate two lots of sub-bass from two parts of the original audio spectrum, and by turning down the dry sound you can adjust the parameters until you get the kind of depth you're looking for. Once I added the processing back into the original kick-drum sound, this gave a welcome impression of extra depth without making the kick drum sound unnatural. I could see this was one plug-in the guys would be playing with some more in the near future, as they were quite enthusiastic about the results achieved.

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Using Analogue Mix Processing With A Computer System

Fader had bought a TLA5060 compressor to use on his final stereo mixes, but to do that he'd been recording the MOTU output, via the compressor, to a stand-alone CD recorder. I explained that, as he had a couple of spare inputs on his MOTU 828, he could send his mix through the compressor, then back into the 828 and route it to a new stereo track in *Logic*. All you have to do is set the record levels, then record the mix directly into the *Logic* Song — remembering not to route this track to *Logic*'s main output while recording, though, otherwise the audio will feed back into itself and cause a howl. Then again, these guys would probably even find a way to use that!

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Overall Compression

Rather than record entire drum tracks, the approach taken by the guys was to record sections (often against a click) to be used to create loops, so toms were rarely used. On a normal session they probably wouldn't have set up the toms at all, and if you don't need them, leaving them out reduces unwanted ringing. The snare sound was deliberately big and ringy, so we went on to try overall compression using *Logic*'s own compressor just to show how different settings can help create different feels.

Conventional settings gave added weight and evened out the sound in a fairly predictable way, but what really seemed to hit the spot was using fast attack, fast release, and a ratio of around 4:1 (hard knee, peak sensing) to make the kit sound pump in a magnificently trashy way. At gain reduction readings of over 10dB, the sound got quite animated in exactly the right way to create exciting loops.

Fader had tried adding reverb to his own recordings, but didn't like the washy sound he was getting. Just to demonstrate the principle, I used *Logic*'s *Platinumverb* set to almost its shortest possible decay time, and with the Balance ER/Reverb setting at around 65 percent early reflections, 35 percent reverb tail. With full brightness, and the bass controls turned down, this gave us a short, bright sound that worked well to add weight and presence but without adding any noticeably cloudy reverb at all. I would have preferred to try some of the *Space Designer* reverbs, but these weren't available in that particular version of *Logic*. Nevertheless, the exercise demonstrated the production value of using a short, bright reverb to liven up a drum sound without diluting the impact.

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Optimising The Computer's Audio Performance

Where possible, you should record and mix at 24-bit resolution, reducing to 16-bit at the last stage in a CD-burning program such as Roxio's *Jam*. However, Fader works at 16-bit throughout so as to maximise track count, and, given the

deliberately grungy, urban nature of his mixes, this is perfectly fine. However, the issue of CPU and disk resources had still been worrying him. His songs are all mixed as audio tracks, and the one he opened by way of illustration turned out to have 41 audio tracks running together. This was enough to occasionally knock the disk's data transfer capacity over the edge and cause glitches, as confirmed by *Logic*'s performance meters. (He was using an external 7200rpm Firewire drive for audio, though tests with the internal drive had produced similar results.) Sensibly, he'd used the track freeze function to conserve CPU power on those tracks that included a lot of processing, but that doesn't help with disk access problems.

However, I noticed that several of his unfrozen tracks comprised very short repeating audio loops and segments, which make the hard drive work much harder than pieces of continuous audio. Freezing the busiest of these (which renders them into a single temporary audio file), got the disk meter reading down to around 70 percent, which I felt rather happier with. The other simple trick is to zoom the screen to show the entire song, so there are no processor-intensive graphic re-draws as the cursor reaches the end of the screen — every little helps.

Both Fader and Kaiser reported that they'd experienced timing problems when composing in *Logic*, and Kaiser, who works mainly on his G4 Powerbook, said that with his Metric Halo 2882 Interface he was hearing too much latency to allow him to record anything unless he monitored only the source while recording. Looking at Fader's *Logic* settings, it turned out that he hadn't switched on the plug-in delay compensation in *Logic*'s Audio preferences menu, and when I checked the same thing in Kaiser's Powerbook, it turned out his was switched off too.



Although a stereo pair of overhead mics was not available, an unmatched pair (Audio-Technica AT4033 & AKG SE300) still gave a very usable sound once the levels were matched.

I still don't understand why this is switched off by default when *Logic* is installed, but it is, so you have to turn it on manually. Once engaged, plug-ins loaded into track insert slots don't throw out the mix timing due to the processing delay within the plug-ins, as the timing of the other tracks is delayed to match. However, be aware that this doesn't apply if plug-ins are inserted into either the buss, master, or aux channels.

In addition, Kaiser's ASIO buffer size was set to 1024 samples, and a setting this large results in audibly high latency. I managed to reduce it to 128 without detriment to his system performance. *Logic* had to be restarted for these settings to take effect, but a quick test with his guitar proved that the latency was now low enough to allow software monitoring.

Fader was using a buffer size of 512, which is good for stability but slightly too high for real-time monitoring or playing of virtual instruments. My usual strategy is to use a buffer size of 128 or 256 when recording, as this gives an adequately low latency, but then as the mix builds up and the CPU or disk access starts to struggle, I set it to a higher value, which gives good stability when mixing and adding effects. Usually you can leave most of the CPU-hungry effects off until you come to mix.

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Comments On The Session

"Anyone who possibly can should try to record some live drums. Paul & Hugh showed us that it was relatively easy, and the results are great, with minimal fuss/experience required on our part. The team also encouraged us to try out recording more stuff (percussion, bongos, claps, ambient vocal 'vibe' tracks) all while someone plays the drums. This added loads to the recordings, and was much better than trying to add it all later. I think worrying about getting miking 'right' puts off too many people, but it's simple really, and once you start, you end up miking up everything to see how it sounds. You just need enough people to make the right noises... Thanks again for all the tips!"



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Practical Recording Tips

We finished the day with a couple more practical questions, both of which were related. Normally Fader records vocals in the same room as the computer, but we felt we could improve both the sound and spill by having the singer stand in the curtained corner furthest from the computer, facing the computer. This would place the cardioid vocal mic with its least sensitive rear axis to the computer, and the curtain would also mop up some reflected sound giving a drier vocal.

Fader also records a classical acoustic guitar in this room, and the same strategy would apply there — set up with the back of the mic directed towards the computer and as far away as possible. He was also unsure as to the best mic position, and though the textbooks, including some of mine, suggest aiming the mic at the point where the neck meets the body, there's no substitute for monitoring through headphones while you move the guitar relative to the mic until you find that magic sweet spot — every guitar is different! One of the small-diaphragm capacitor mics would be best for this application.

The other related issue is that of computer noise, and it is possible to quieten G4s to some extent by putting a folded towel or rug over them, leaving the front and rear open but covering the top and sides right down to the floor. This doesn't obstruct the airflow and so shouldn't cause any heating problems. Also, hanging a folded towel in front of the computer, but again leaving room for the air to circulate, reduces the noise level noticeably, especially the high-frequency 'whine' components. Finally, as the back of the shelf holding the computer was open, with a hard plaster wall behind, we suggested hanging foam or heavy fabric on the wall itself to cut down on reflected sound from the rear of the machine.



With the kick-drum sound still leaving something to be desired, Paul turned to some of *Logic's* plug-ins to see whether judicious processing could sort it out.

That about wrapped it up for our visit too, and as the biscuits had all but gone we decided it was time to head for home! **SOS**

Published in SOS January 2005

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Studio SOS

Glenn Bucci

Published in SOS January 2004

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People : Studio SOS

This month, the SOS team are in New York helping reader Glenn Bucci with his recorded guitar and bass sounds.

Paul White

Reflections from the alcove to the right of the main studio workstation were causing monitoring problems, so Paul and Hugh nipped down to the local music shop to get some acoustic foam. This was applied to the most problematic wall, taking care to cover the corner as well, and this gave a significant improvement in the monitoring accuracy.

Glenn Bucci, an SOS Web Forum regular, got in touch with us to see if we could help sort out a few mixing problems he was having in his basement studio, located around an hour from New York in the town of Huntington, just beyond the notorious Amityville — yes, it is a real place! Apparently Glenn was a regular follower of our Studio SOS exploits, as his wife unveiled a massive plate of chocolate biscuits almost as soon as we arrived!

Glenn's studio is based around a PC running *Cubase SX* running under Windows XP, though he had only 384MB of RAM, which was barely significant for his application, and we suffered occasional glitching during our visit that may have been due to this. However, as Martin Walker is not an allowable item of hand luggage, and because I have very little experience of *Cubase SX*, we didn't look into this any further, especially as Glenn had promised himself a new and faster computer in the near future. An RME card with ADAT I/O served as the audio interface, communicating with a Behringer DDX3216 digital mixer fitted with a pair of ADAT I/O cards. A pair of Mackie UAD1 cards in the PC provided extra processing power. Glenn's mics included a Rode NT1A, a Blue Blueberry, an Audio-Technica AT4033 and an Electrovoice EV237.

On the hardware side, Glenn's Manley Laboratories Langevin Dual Vocal Combo was his most recent purchase and favourite input device, though he had a

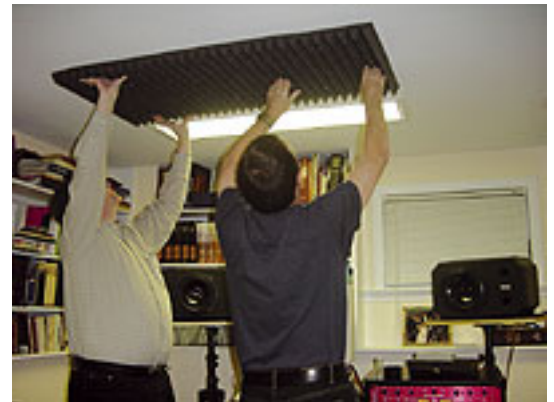
Focusrite Voicemaster for when more inputs were needed, in addition to Focusrite Compounder and Mix Master units. He also had some very nice guitars, a Mesa Boogie V-Twin recording preamp, a Fender Twin and a 12W Marshall amplifier. The monitoring comprised a pair of Tannoy 800s supported on adjustable tripod stands, with home-made wooden top plates and ribbed rubber mats to provide a secure and vibration-free fixing. These speakers have a fairly wide dispersion, which is a good thing generally, but in this case their potential was being compromised to some extent because of the room geometry.

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Improving The Room

The studio room turned out to be a slightly awkward 'L' shape, because a large cupboard for the boiler (or furnace as our US friends prefer to call it) projects into the room. Luckily, Glenn has not found noise from the boiler to be much of an issue, since most of the instruments go direct into his DAW. Also, during the cold months when it is in use, he can turn it off temporarily during a vocal or acoustic guitar take, and his well-insulated basement usually maintains a comfortable temperature.

The equipment and monitoring was all set up to face down the long axis of the room. The alcove to the right of the mixing position seemed destined to cause acoustic problems with sound bouncing back to the listening position, so we decided to drive down to the local Sam Ash (a chain of music shops across America) to buy a few panels of Auralex foam to treat this. Although professional acoustic engineers may cringe at these foam-based 'quick fixes', the reality is that they can make a significant and very cost-effective improvement in the home studio, even though they are unlikely to bring the acoustics anywhere close to professional standards. As the photographs show, we covered most of one alcove wall facing the front of the studio to try to minimise any reflections from that surface bouncing back to the listening position via the front wall.



Another foam panel was also applied to the rather low ceiling between the monitoring position and the monitors.

The corner of the wall was also a potential problem, so we were careful to wrap the foam around the corner to soak up any high frequency reflections from that area. By aligning one of the notches in the foam with the corner edge, the foam folded neatly around the corner — although we allowed the main section of panel to stick firmly before trying to fold the shorter section around the corner. A further panel was fixed to the rather low ceiling in the traditional position, between the monitors and the listening position. The ceiling was low, because Glenn had specified a high degree of sound insulation in the joist space to minimise noise to

and from the family room upstairs.

The outcome of our acoustic treatment was a slight but noticeable tightening of the stereo image and, as a bonus, the dead wall in the alcove would be useful when recording vocals, as the singer could stand with their back to the foam to help provide a more acoustically dry recording.

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Recording Acoustic Guitar

Glenn played us some of the tracks he'd been working on, and straightaway we recognised some of the problems he'd described — as well as a couple he hadn't! The first song featured nylon-string classical guitar, which Glenn had compressed, EQ'd and added quite a lot of reverb to. He wasn't initially too unhappy with the sound, but I felt that we could help him improve on it, especially as he had some nice mics, as well as the Manley preamp.

After bypassing all his processing, the guitar track sounded somewhat congested and 'pumped' on occasions, so we asked Glenn if he'd compressed it during the course of recording. He told us that he'd applied just gentle compression to prevent overloads during the recording. However, the effect was very obvious to the point of being detrimental, and it couldn't be undone by any further processing, so we suggested that he re-record the part from scratch using no compression, so that we could go through each stage of the process with him. As he was recording everything with 24-bit word lengths, we could afford to leave 8dB or so of headroom without losing any significant resolution — which in turn meant we didn't need compression as a 'safety net'.

We set up the mic (an Audio-Technica AT4033, which is a cardioid back-electret capacitor design with a medium-diameter diaphragm) over a piece of plywood placed on the floor — Glenn had got this idea after reading about the technique in a previous issue of *Sound On Sound*. The idea of the board is to reflect some of the sound



Listening carefully to some of Glenn's recordings revealed that he had been processing the acoustic guitar fairly heavily at the mixing stage. With the mix processing bypassed, Paul felt that there were inherent problems in the original recording technique, so he had Glenn set up as usual for recording (right). As suspected, Glenn's mic position, two feet in front of the sound hole, was creating a boomy sound which Glenn had been trying to correct with processing.

back up from the floor in order to give a more lively sound when recording acoustic guitars and similar instruments. The mic's suspension shockmount uses an elasticated cord, providing isolation between the inner and outer frames, which is threaded through holes in the inner frame and hung on hooks on the outer frame, above and below. After 12 years of use, the elastic had stretched well past the point of no return and consequently not only provided a complete lack of shock isolation, but also fell off the suspension hooks with the slightest provocation, putting the microphone at risk from the effects of gravity!

Replacement suspensions can be purchased from Audio-Technica, but it's fairly easy to replace the elastic yourself — suitable elasticated cord can be purchased from any haberdashery shop, and the lacing is fairly self evident. The original design uses a metal crimp to secure the ends of the cord, but a simple knot will suffice. The critical thing is to get the length and tension of each support loop right, and that takes a little trial and error, but the results are very worthwhile. If you find yourself repairing a similar shockmount, pay attention to the condition of the rubber bands which hold the microphone in place as well, as these tend to dry out and go brittle with time, and then they split easily.

Placing the AT4033 level with the guitar and aiming it at the point where the neck joined the body gave a fairly natural sound over the headphones. I made some further adjustments while listening to Glenn play and suggested using a higher mic position to lose some of the low end from the sound. Glenn was a bit sceptical of this, so we didn't try it for the first test recording, but as soon as the recording was complete he felt the need to add some high-end EQ within *Cubase* to brighten the sound. To avoid being over-reliant on EQ, we suggested recording again, but with the mic raised slightly higher above the neck of the guitar, so that the sound was closer to what he wanted at source. This would avoid having to use so much EQ, essentially to correct an inappropriate mic position.

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Guitar Processing

The resulting recording was nice and bright, with less muddiness at the low end, and it sat in the mix much better, so we went on to try to establish some suitable compression and reverb settings. This time we used the Urei 1176 compressor simulation plug-in from Glenn's Mackie UAD1 card to apply fairly gentle compression. We used a faster release time than I suspect Glenn had used before, so that the compressor would have time to recover between notes. This worked fine with just three of four decibels of gain reduction showing on the peaks, so we moved on to fixing the reverb.

Originally Glenn had used the UAD1 *Dreamverb* plug-in to set up a church-like acoustic, but both Hugh and I felt that this was swamping the guitar sound, and also robbing the mix of much of its valuable space. Instead, we set up a shorter, brighter room sound that complemented the attack of the guitar and created a

nice impression of space without washing out the sound. The exact settings we arrived at can be seen in the screenshot, though I've no doubt we could have improved on this further had we had more prior experience with this particular reverb plug-in. Glenn's drum part, which was sequenced using a Steinberg *LM4* virtual instrument, was also struggling against overwhelming reverb, so we used the same small bright room and backed off the level to get a more natural sound.



Once a more satisfactory guitar recording had been achieved, the EQ and compression settings could be kept quite subtle, as can be seen in the screenshots above.

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Glenn's Comments On The Session



"First I want to say it was great to meet the staff of *SOS* — I had a lot of fun! Having now listened to the effects of putting up the acoustic foam, I can confirm that my stereo image has indeed improved, and I know this will help me get more accurate mixes in the future. As for the shockmount, I've now repaired it, and it works and looks better.

"I was pleasantly surprised by Paul's mic placement on the guitar, as I normally just put it a couple of feet in front of the instrument. It seems it really is worth spending the time to get the

correct mic placement to capture the sound you want, rather than using an EQ to correct the sound later. Paul and Hugh also taught me that, while getting a great sound on your bass is important, you also have to find the correct sound to suit the type of music you are doing. My smooth jazz bass sound wasn't right for a blues number, but Paul's punchier, more mid-range sound was just what the doctor ordered!

"With 24-bit recording, Hugh emphasised that leaving enough headroom to avoid clipping would not degrade the sound, as the final track would be converted to 16-bit resolution for CD use anyway. Leaving 8dB of headroom is usually better than pushing your signal close to the red and then using compression or limiting to avoid clipping. I also learned that adjusting (reducing) the low-end EQ on a reverb can help take the boxiness out of the sound. When I recorded my acoustic, I was trying to get the same reverb Larry Carlton did on his recording of the Lords Prayer (in the late '80s/early '90s, mind you). However, I actually preferred the more natural sound that Paul came up with, which sounds like the guitar is in front of you, not at a church recital. Thanks again to the SOS team — I really enjoyed meeting up, and I learned a lot."

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DI'd Bass Guitar

We then went on to tackle Glenn's bass guitar sound, so he opened up a cover version of Eric Clapton's 'Stepping Out' he'd been working on (originally from John Mayall's Bluesbreakers' *Beano* album), where he'd simply DI'd his bass via the Manley preamp, while adding some compression. There was nothing intrinsically wrong with the bass sound, though it lacked mid-range definition, which meant it wasn't being heard as well as it might have been when the rest of the mix was up and running. We suggested re-recording the part using his Mesa Boogie V-Twin tube preamp, which we then fed into the Manley preamp so that

we could use its compression facilities if we needed to. We used the V-Twin's clean channel with Gain set at around 25 percent, Bass set halfway, and Mid at 90 percent. Treble was set low at 10 percent, while Presence was at 30 percent. These settings gave a nice crisp sound with plenty of mid-range definition that would really cut through in the mix.

We actually used very little compression while recording, and probably needn't have used any at all, but it still sounded DI'd to me, so we agreed to try some 1176 compression to give the sound a bit more attitude. We also added a little EQ boost at 80Hz and 250Hz (using the UAD1 *Cambridge EQ* plug-in) coupled with some gentle HF roll-off, giving us a sound that sat rather better in the mix. As a final tweak, Glenn inserted a Steinberg tube-simulation plug-in and adjusted the drive until the sound was just starting to roughen up, which gave more of an amp'd feel. One tip here is that, if you do use a tube simulation plug-in and you feel it is making the sound too edgy or gritty, you can put a sharp high-cut filter after, with a slope of, say, 18dB/octave at 5kHz. Because every bass sound is different, you should try adjusting the filter frequency so that the sound smooths out, without actually becoming dull. Though these various tricks and adjustments gave us more of an 'amp' sound, a dedicated processor, such as a Line 6 Bass Pod XT, would make this very much easier and offer a far greater choice of tones.



Here you can see the settings which Paul used for adding a short natural reverb to Glenn's guitar sound.

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Recommendations

Glenn's monitoring setup actually worked pretty well, so a minimal amount of basic treatment was all that was needed. He has been careful to choose good-quality equipment, both hardware and software, so the main mix problems he was encountering were related to overprocessing, specifically with compression and reverb. Again, our visit underlined the importance of getting as close to the desired sound as possible at source, both by choosing a suitable mic and by adjusting its position, as this minimises the amount of EQ needed and invariably leads to a more natural and open sound. It's also important to make the final EQ adjustment with all tracks playing, as the subjective result can be



Glenn's bass sound had been disappearing in one of his mixes, so Paul suggested recording through a Mesa V-Twin valve recording preamp in order to add some mid-range definition and general attitude.

quite different to EQ'ing in isolation.

The UAD1 1176 compressor proved very effective on the bass and guitar tracks that we worked on, and setting up a brighter, more intimate reverb helped retain clarity and create a sense of space. However, compressing for no good reason (or over-compressing) can seriously compromise some sounds, and is best avoided by recording without compression where possible and then trying various compression settings afterwards at the mixing stage. The downside of over-compression was also evident on a sampled piano part Glenn had recorded, where the compression made it sound somewhat hard and electronic. As a rule, I wouldn't compress acoustic pianos (or good piano samples) at all.

Glenn is a big fan of '80s productions where reverb was used more extensively than it is now, but to get away with that type of production you need to use a very high-quality reverb unit or plug-in and, equally importantly, choose a setting that doesn't muddy your mix. *Dreamverb* was clearly up to the task, but it meant doing some patch editing. Rolling some of the low end out of the reverb is one way to avoid mix congestion, but getting the effect exactly right takes some experimentation, and it's worth saving any good patches you come up with. Adding 60-80ms of pre-delay can also help when you're treating vocals, because this provides some separation between the original sound and the reverb. Glenn had originally chosen church-type reverb patches, which are characterised by a lot of rolling low end and very little high end. While the UAD1 *Dreamverb* produces extremely convincing church reverbs, these weren't really suitable for the type of music being worked on. As a rule, smaller, brighter rooms, plates or ambience settings create the required sense of space and complexity without obscuring the sound or compromising its dynamics.

Finally, we felt Glenn would benefit from dual flat TFT screens on his computer, as these would help manage the windows in *Cubase* better and also help reduce the amount of hum on his Fender Stratocaster guitar due to magnetic radiation. When Glenn played his Strat, there was a further interference problem that we couldn't track down in the time available (we had an appointment with a curry downtown!), but during the drive back to Manhattan we concluded that it could have been radiation from the rather large TV set in the room above. The only solution is to turn off such potential sources of interference while recording and/or fit the guitar with good noise-cancelling replacement pickups (such as those made by the Australian company Kinman) that won't compromise the tone. **SOS**



With the preliminary recording problems out of the way, the SOS team faced up to the day's main challenge...

Published in SOS January 2004

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SOUND ON SOUND

The World's Best Music Recording Magazine

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Studio SOS

David Ashman

Published in SOS February 2004

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David Ashman felt that his mixes lacked energy and were sounding 'too digital', so the SOS team set off to his home in Bristol to sort out his monitoring system and mix processing.

Paul White

David Ashman called us in because he was having trouble getting any energy into his mixes, and he also described them as sounding 'too digital'. However, he also thought his monitoring environment might be giving him problems, so we set out for his Bristol home studio to see if we could help. As David was a regular reader of the Studio SOS series, he was aware of the unwritten rules, and so had coffee, chocolate digestive biscuits and chocolate cake prepared in readiness!

David's studio is now centred around Emagic's *Logic Audio* running on a rather noisy dual-processor Mac G4 and using an Emagic EMI 2|6 USB interface, though his previous musical experience had been based on Sonic Foundry's *Acid* running on a PC (which he still has). The EMI interface was powered up via USB, and we had to boot up the computer twice before we got any audio. In theory USB power should be OK, but as the interface is also designed to be able to work from an external PSU, I felt that getting a suitable PSU and powering it from the mains might be safer.



Because the monitors were set up along the long wall of David's narrow studio room, the bare wall directly behind the mixing position was causing monitoring problems. A duvet slung between mic stands over this area created an immediate improvement in the sound, so it was suggested that David mount a rail on the wall to allow a duvet to be hung there more permanently.

Dealing With The Room

The bedroom housing the studio is long and narrow, with the speakers (passive Spirit Absolute 2s) set up on stands along the longer wall and spaced rather too widely for the listening distance. The surfaces of the room were completely untreated and painted bright green, so that when Hugh was taking photographs I felt almost as though I was on a green-screen *Star Wars* set!

We always start out by listening to some original mixes in the room prior to making any changes and suggestions, and it immediately became evident that there was more than one problem, the most troubling being the inaccuracy of the monitoring system. The choice of sounds used in the mix was also questionable, though without accurate monitoring, choosing the best sounds is very difficult. We've come across Spirit Absolute 2s in this series before, and again we found that they were simply not telling us what was going on in the critical lower octaves, though, being passive, the choice of amplifier used to drive them would also have had an effect.



Once the rear-wall reflections had been tamed, critical listening to David's mixes highlighted the ineffectiveness of his monitoring system at low frequencies, a deficiency which had led him to make unsuitable choices of instrumentation and made mixing very hit and miss.

Checking David's mixes with Hugh's rather nice Sony MDR7509 headphones confirmed that there was quite a lot going on at the low end that we simply weren't hearing. There was no practical way to fix this using the Absolute 2s so we recommended David invest in a full-range pair of active monitors. We also suggested that he move them closer together, possibly by moving both his Mac and PC systems into a single computer desk rather than having the speakers separated by two computer desks. David also had to do a lot of unplugging to play back mixes, so in the absence of a hardware mixer, a monitor controller such as the Samson C*Control would be ideal, as this also handles the level control for active monitors, provides multiple source switching and includes headphone monitoring and talkback.

Because the wall behind the mixing chair was completely bare, we tried the old duvet trick to help soak up some of the reflected sound, with a view to tightening up the stereo image and reducing coloration. As luck would have it, the rear wall was lightweight plasterboard on a studding frame, which meant it would absorb or allow through a lot of the low end, leaving the duvet only to soak up the mid-range and high frequencies. This meant the duvet was more effective than if the wall had been solid brick. Having proved the principle, David said he'd fix up a rail from which he could hang a duvet more permanently. Using a rail to hang the duvet a few inches from the wall is more effective than pinning it directly to the

wall.

Because the side walls were a fair distance from the monitors, and because one was largely taken up by a window, we didn't advocate applying acoustic foam to them. Instead we suggested that the bare wall opposite the window (to the left of the monitoring position) be used to house a bookshelf or something similar, just to break up side-to-side reflections. That was about as much as we could do on the monitoring front, though I invited David along to my studio so that he could compare some different monitors, including my Mackie HR824s and whatever else was in for review at the time.

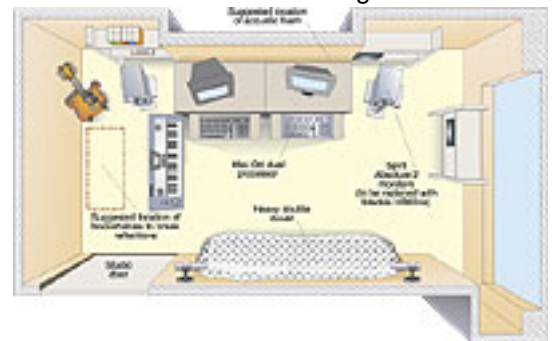
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Problems With Over-processing

Getting back to the mixes, we thought the bass sounds were rather bland and lacking in punch, so we asked David to open up the original *Logic* song to see what he'd done. It turned out that David had not yet rigged up a MIDI interface to his new Logic system, so all his parts had been written in step time, giving them a somewhat mechanical feel. Most of the sound sources came from samples loaded into the *EXS24* software sampler and, rather than optimising the track levels, David had inserted a limiter plug-in into each track to prevent the signals peaking. This worked quite well on the drum tracks, but was unnecessary on the other parts. Having said that, limiters can be useful on resonant synth parts, as some of those resonant peaks can be very loud. He'd also used *Logic's DJ EQ* plug-in on most tracks, along with compression, but again these proved largely unnecessary. Where EQ is needed, the new *Channel EQ* is more flexible and sounds good.

For reverb, David had used four instances of Audio Ease *Altiverb* set up on four different send busses. These took up a very significant proportion of the available CPU resources, and we felt the use of reverb was rather heavier than the track needed, which robbed it of some of its impact. As an experiment, I used a single *Logic Platinumverb* set to a short, bright, plate-like setting. It may not have been as refined or realistic as the *Altiverb*, but it sounded more appropriate in this instance, and the CPU load dropped to almost nil. David had also bought a second-hand TC Powercore PCI card, but had not yet installed it, so I suggested that he try the excellent *Classicverb*, which is bundled with the card, when he got it running.

Diagrams: Tom Flint



Here you can see the layout of David's studio, showing the overly wide spacing of the monitor speakers with regard to the listening position, which was compromising the stereo imaging. A narrower desk setup for the two computer workstations would allow the monitors to be placed closer together. Also note the suggested positioning of the double duvet (using a wall-mounted rail to provide a little clearance between the duvet and the wall) and the recommended addition of some bookshelves at the left-

David had also used the default *EXS24* sine wave for one of the parts, which is fair enough, but he'd also added EQ and compression. MIDI parts can often be controlled using *Logic*'s MIDI dynamics rather than compression, though compression can be effective on some drum or percussive sounds. However, using EQ on a sine wave is not very useful, as EQ can only change the level of a sine wave, not its harmonic content, because it has none! David had learned the theory behind this, but had left the EQ switched in by default. Instead, to make the sine sound more interesting, we tried the *Logic Phase Distortion* plug-in, which turned out to be remarkably effective and gave the track an organ-like quality with a hint of FM edge. A tempo-related *Tape Delay* plug-in was added to a sequenced piano sample to give it more of an arpeggio feel.

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Bass & Drum Tweaks

As the bass parts were the obvious weak points, we isolated them and listened to the two layered parts David had created in isolation. I suggested that one of the parts could be replaced using the *Logic ESM* monosynth plug-in, as it is capable of very TB303-like bass sounds that I felt might work well in the track. Judging the effectiveness of this new bass sound was very difficult over the Absolute 2 monitors, which were again telling us there was no low end, though Hugh's headphones confirmed that there was plenty of bass! For the other bass part, which used a synth sample, we again processed it using a moderate amount of phase distortion, just to dirty it up a little.

David had also layered two drum parts, one of which had been treated with a flanger. We tried *Logic*'s *EVOC20* filter bank, which we activated in time-limited demo mode, in place of the flanger and used a tempo-locked LFO to morph between two filter settings to create a fluid, rhythmic sound that really emphasised the higher percussive parts. The movement of the modulated filter added interest to what was otherwise a very straightforward drum part.



Passing one of a pair of layered drum parts through *Logic*'s *EVOC20* vocoder, and modulating it between two different filter-bank settings, helped to add interest to an otherwise repetitive part.

David had also dosed his drum parts with a generous helping of *Altiverb*, but, because the parts were not split over several tracks, all the drums got the same treatment. This invariably makes the kick drum too reverberant, so the ideal solution is to split the drum part over two or more tracks, with the kick on its own track. However, it is possible find a suitable reverb to work on a whole drum part, provided that you roll off the low end of the reverb sound. You can do this using the filtering parameters in the reverb itself, or by inserting a low-cut (high-

pass) filter before the reverb, set to between 150Hz and 200Hz. This filtering, combined with a short reverb of between 0.6s and 1s, can work well when you have no way to split up the drum part, and in any event you only need enough reverb to take the dry edge off the sampled drum sounds (in this case GM drum kit samples).

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Getting Rid Of The Mechanical Feel

At a purely artistic level, I felt that writing everything in step time using *Logic's* Matrix Edit window and then copying blocks gave the drum part a somewhat sterile feel, so I added a few hi-hat beats (again in the Matrix Edit) which varied from bar to bar over a four-bar cycle, just to demonstrate that small variations can make a part sound more organic. David admitted that programming good drum parts with the right groove was quite difficult, so in addition to the more obvious drum loop sample CDs, I suggested he try some of the Keyfax Twiddly Bits MIDI drum loops, as they are inexpensive and can be used to trigger any drum sounds. David liked this idea, because he was worried that if he used sampled loops, these would be recognisable.

We also listened to some of the more creative loop-based music that David had made in *Acid*, much of it incorporating sounds he'd recorded, often via a simple binaural mic setup. Of course the spatial magic of binaural recordings only works properly over headphones, but the overall effect was powerful and imaginative. Because David was obviously comfortable working this way, we felt he might try creating some rhythmic parts in *Acid* using the PC, then save these as stereo WAV files to be imported into *Logic*, where they could be chopped up, copied and looped to be used as the basis for new compositions, rather than relying on off-the-shelf loops. I'd also have liked to have heard some tracks with David's guitar more in evidence (which he records via a Line 6 Pod Pro) as just a track or two of a 'real' instrument can add a lot of life and depth to an otherwise all-MIDI composition.



Here you can see how *Logic's* Multipressor plug-in was set up to subtly enhance David's mixes. The number of bands was reduced to three, and low-threshold, low-ratio compression was applied in each band. In addition, the make-up gain for the middle band was reduced by half a decibel to create a slight 'smile curve'.

Another processing technique we experimented with was using *Logic's* Tremolo plug-in as a chopper/panner by setting the wave shape to square and setting the two outputs to opposite phases of modulation. In other words, when the left side was turned up, the right side was turned down, and *vice versa*. By using the tempo sync function to chop 16 times every bar, interesting rhythmic modulations can be created that work well on pad and even vocal parts — although if you

were to use this on vocals, it would be best confined to short passages.

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Rap Vocal Processing

Next we turned our attention to a rap vocal part that David had recorded using a young vocalist who'd actually performed in the same room as the computer, and monitored over speakers rather than headphones. Consequently there was some spill, but as the vocalist had a fairly strong voice, this didn't cause too many problems. To even up the level, I used *Logic's Compressor* plug-in with a ratio of 6:1 and a hard knee, adjusting the threshold to give around 6-8dB of gain reduction on the loudest phrases. The attack time was set as fast as possible, with about a quarter of a second of release. The rather overbearing *Altiverb* was swapped for a short, bright *Logic Platinumverb* to create a more intimate, 'in your face' sound, though, given time, I'm sure that we could have called up an *Altiverb* ambience program that would have done the job just as well, and with a better sense of real space.

David had actually recorded the vocal directly via a Digitech Vocalist Workstation vocal processor, but as he also had a very competent TLA D1 valve mic preamp lying around doing nothing, we felt he might get better results if he recorded using that, adding any necessary vocal processing while mixing. In fact I think it's a good general rule to record as much as possible with no EQ and no processing, at least until you gain more experience, as you can always go back to a clean slate if the mix is proving difficult.

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David's Comments

"Since your visit I have taken on board your suggestions, and ruthless decisions have been made! The Absolute 2s are in the SOS Readers Ads — although I hope I've sold them before anybody reads this article! As soon as they are gone I will purchase a pair of Mackie HR624's, and I will also implement the acoustic treatment suggestions you made. I have also ordered a pair of Sony MDR7506 headphones. I now have the Powercore card installed, and it has eased my CPU problems — I can confirm that the plug-ins sound great, especially the reverb. The only problem I can see is that these DSP cards are going to become addictive!

"I have also put into practice your mixing, mastering and songwriting tips, and I feel they have given me a deeper understanding of what I should be looking out for in the future. Finally, I would like to thank Paul & Hugh for visiting my humble studio and for their professional contributions regarding my problems — I have gained much insight as a result."



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Mastering In Software

David's final question related to mastering, as he wasn't sure whether to buy dedicated hardware or to rely on plug-ins. After David first contacted SOS, Reviews Editor Mike Senior had processed one of his earlier recordings using a Drawmer DC2476, which opened up the sound and revealed high-end detail that had been lost in the original mix. Again the monitors didn't tell us how well this worked at the low end, so we double-checked using Hugh's headphones.

Because David didn't want to pay out for hardware if he could get an acceptable result using software, (and we'd already added new monitors and headphones to his shopping list!), I tried the simple approach of putting *Logic's Multipressor* multi-band compressor in the main stereo mix insert point, followed by the *Limiter*. I set the compressor to work over three bands, rather than the default four bands (just to make it more manageable) and set the crossover points at around 120Hz and 5kHz, so as to keep the critical mid-range intact. The compression ratio in each band was set to around 1.2:1 and then the threshold was adjusted to give just a few decibels of gain reduction in each band. On most mixes, this means having a threshold setting of between -30 and -40dB. Using a slightly higher ratio on the bass band can increase the low-end density.

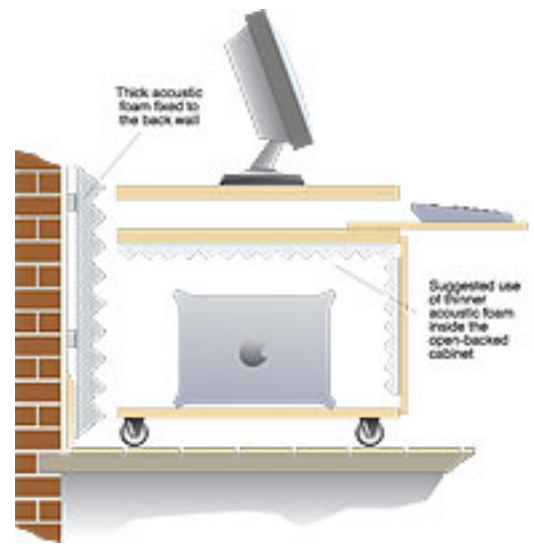
This approach to compression provides a homogenising effect by compressing gently over a wide dynamic range. This is different to the way individual tracks tend to be treated, where it is normal to set the threshold much higher and then use a higher ratio so that only excessively loud peaks get stamped on. The tonality of the mix can be balanced by altering the relative levels of the three compressor bands, so if you want the 'smile curve' loudness effect, all you need do is pull down the mid-band by a couple of decibels relative to the high and low ends. The limiter takes care of any signal peaks, and the threshold should be set so the gain reduction is no more than around 4-5dB on the signal peaks. I have to admit that *Logic's Limiter* puzzles me, because when you set it up so that you'd think it was limiting everything at around 0dB, the output peaks at around -5dB, and it doesn't have an output target control where you can determine the exact value at which limiting occurs. For my own work, I always use the Waves *L1* when I need limiting, which I still find to be the most useful and intuitive of all the limiter plug-ins I've tried, though I know there are some equally well-specified contenders from the likes of TC Electronic.

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Tackling A Noisy Computer

Before wrapping up, we decided to have a go at quietening the extraordinarily noisy 'mirror door' Mac G4, which was sitting in the bottom of a lightweight, open-backed computer cabinet with the door open. Closing the doors didn't help much,

as most of the noise seemed to be coming out of the back, so we moved the cabinet away from the wall slightly and redeployed the duvet behind it just to see how much of a difference it just made. The improvement wasn't dramatic, but it was noticeable, and now closing the doors did make an improvement. Without wanting to specify anything too elaborate, we settled on suggesting acoustic foam on the wall behind the cabinet, with thinner foam used inside the cabinet and door to kill reflections inside the box. This should improve the noise situation by a few welcome decibels, while still allowing air to circulate around the computer for cooling.



Paul suggested reducing the noise from David's PC by placing thick acoustic foam on the rear wall to absorb sound emerging from the open back of the cabinet, as well as using thinner foam on the inside surfaces of the cabinet.

For critical vocal parts, David could run a mic cable out onto the landing, where the computer was to all intents and purposes inaudible. Although the landing was long and tended to colour the sound, experiments with the ubiquitous duvet confirmed that hanging one across the landing to isolate the end where the recording was being made improved the situation considerably, though hanging another duvet over the wall and door at the end of the landing that the singer would be facing made the sound cleaner still.

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Recommendations

Yet again we found that what was originally described as one problem had numerous facets, the most serious of which was the choice of monitors. We've come across Absolute 2s twice in the course of our SOS visits, and though they are fine in the mid-range and at the high end, we've found them to be completely inadequate at the bass end, leaving the user guessing as to what's going on down there. They are fine as secondary monitors to tell you how something might sound on a small domestic hi-fi, but I can't recommend them as main monitors. Consequently, replacing these with something more honest should be a first priority, and improving the monitor geometry by reducing the spacing between the speakers should also be high on the list.

The amount of room treatment needed is pretty minimal, and a heavy double duvet across the back wall should fix the most serious problems, as the partition wall behind it will work as an impromptu bass trap. I must stress that the duvet trick is just a cheap and cheerful fix, and doesn't give you the same results as a properly designed studio, but from a pragmatic viewpoint such simple fixes are usually enough to give you an acoustic environment you can work in without the

sound being too misleading.

As David is a keyboard player (as well as a guitarist), we feel he should get a MIDI interface up and running as soon as possible, so that he can use real-time feel and expression, rather than adding notes in step time. Drum parts can be livened up using sampled loops, MIDI drum grooves, or parts he creates for himself in *Acid*, and once the monitoring has been sorted out it should be much easier to choose sounds that work properly in the mix, rather than relying on processing to try to re-shape things later. **SOS**

Published in SOS February 2004

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SOUND ON SOUND

The World's Best Music Recording Magazine

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Studio SOS

Nigel Helm-Nurney & Pombokiwi

Published in SOS March 2004

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People : Studio SOS

This month the intrepid SOS team travel to Nigel Helm-Nurney's London studio to help his band Pombokiwi do battle with unwanted spill, uninspiring guitar sounds, and masses of egg boxes!

Paul White

This month's column saw us back in north London after Nigel Helm-Nurney called us to help out with recording his band Pombokiwi, which comprises himself, Etienne Baird, Simon Davies and Howie Hughes — although Howie was unable to join us for the session. Nigel plays acoustic guitar, Simon electric guitar, Etienne takes on vocals, and Howie plays the same model of Roland V-Drum kit as I have in my own studio. Pombokiwi's studio is a rather well-converted garage at the rear of singer Etienne's house, where studding walls, plasterboard, acres of dense Rockwool, double-glazed windows and a double door have created a fairly soundproof working environment. The project didn't take long, as all the band members are involved in the building trade in various capacities!

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Inside The Studio

The inside of the studio is festooned with fabric, which helps tame any ringing, but prior to contacting us the band had bought a huge stack of egg boxes to use as acoustic treatment. They'd made a start sticking these to the walls, but called a halt when we suggested they'd make very little difference, especially with all that fabric hanging over them. For the record,

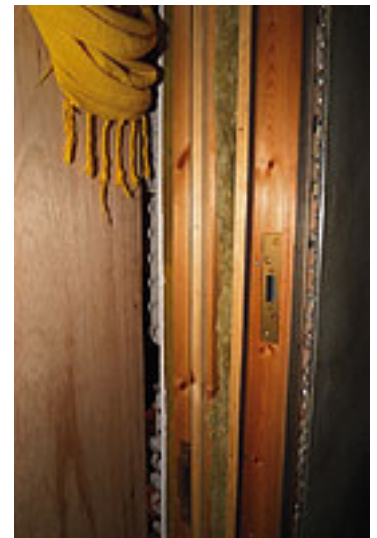


Although the track limitations of the band's Tascam 788 multitracker restricted their recording options to a certain extent, they also weren't using its facilities in the most effective way.

egg boxes do produce a little useful scattering at higher frequencies, but they have negligible soundproofing benefits and far better acoustic improvements can be achieved using drapes and/or foam panels. Where did that egg-box myth come from? Nevertheless, the room did sound very usable, with reasonably well-controlled reverberation across the spectrum and no obvious bass problems other than the choice and location of monitors.

The recording system comprises a Tascam 788 eight-track digital recording workstation augmented by a Mackie CFX16 mixer and a handful of microphones, including an Audio-Technica AT3035 large-diaphragm capacitor model and the ubiquitous Shure Beta 58. For monitoring, the band has a small pair of Sony hi-fi speakers (part of a compact Sony music system) and a pair of moulded PA speakers, one of which was hidden behind the drapes, plus there was a powered wedge monitor for foldback duties. They had been using these PA speakers as their main monitors, but as playing back some music soon proved, these produced a hugely coloured, bass-heavy sound, in part due to them resting on the floor close to the room's corners. Even correctly placed, they were not going to be anywhere near accurate enough to use as studio monitors.

The little Sony speakers fared somewhat better, but were not really up to the job of serious monitoring, as they had no real bass extension and were voiced to flatter a budget music centre rather than to be accurate, so Hugh and I suggested that upgrading the monitoring system should be made a priority. The speaker position also needed changing, as the monitors were set up much too far apart. As the Tascam recorder and Mackie mixer were set up side by side on a narrow bench at the centre of the long side of the room, placing the monitors either side of this control centre would be optimum, though raising them above the bench rather than resting them directly on it would give the best results, as it would help avoid reflections from the bench surface.



The band had made good use of Rockwool between the outer wall of their garage and the inner plasterboard wall, and this had soundproofed the room quite well. However, when it came to acoustic treatment in the room, they had fallen for the egg-box myth...

Because of the style of music, the band like to record everything at once, including vocals, but this was causing problems with spill into the vocal mic, a nice Audio-Technica AE5400 hand-held vocal condenser mic. Nigel was using a Carlsbro acoustic guitar amp, which he was then DI'ing from the effects send socket, while Simon was using a small Marshall combo, also DI'd via it's rear-panel speaker-emulated DI jack. The drum kit, being electronic, was plugged directly into the recorder (as a mono signal) via the Mackie mixer. The Mackie insert jack trick was used to provide direct outs from the guitar amps, drum kit and mic to feed the recorder. By plugging a mono jack plug only halfway into the insert socket, a direct output can be taken without disrupting the signal path through the mixer.

The PA speaker was used to provide foldback to the drummer, while the floor wedge provided foldback for the vocalist. However, not only did this setup cause spill problems, there was also a nasty ground-loop hum which we traced to Nigel's DI feed from his acoustic combo. This could have been cured by putting a DI box in series, but as neither Nigel nor Simon were completely happy with their guitar sounds, we decided to try to find an alternative recording method. But first we took a look at the spill problem.

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Reducing Spill & Improving The Vocal Sound

Because everything was being DI'd, both Hugh and I concluded that a multi-output headphone amplifier fed from the recorder's single headphone outlet would provide the most practical solution, enabling each of the band's members to wear headphones for monitoring. This solution would mean that they would all hear the same balance, but would at least be able to adjust their individual monitoring levels. The inaccurate PA speaker and the wedge monitor could then be switched off, thus preventing spill from those sources, and the headphone mix would be much closer to the actual sound being recorded. Numerous low-cost headphone distribution systems are available from companies such as Samson and Behringer, so this is not an expensive solution. A further possible refinement would be to make use of the V-Drum kit's external audio input, so if the drummer required a different balance, this input could be fed from one of the mixer's pre-fade sends to provide a custom balance of instruments and drums via the drum 'brain' module's own headphone amp.

Although there would still be acoustic guitar playing and some audible tapping on the drum kit, judicious mic placement would be able to reduce their spill on to the vocal mic to an acceptable level, something we proved by setting up the vocal mic at one end of the room facing a corner with Etienne singing towards the rest of the band. However, we felt that, for the best results, he should invest in a thick king-size duvet (whatever would we do without them!) and hang it from the ceiling about half way down the room to form a semicircular enclosure. He could then stand with his back to the duvet when recording, keeping him far enough from the walls to avoid coloration and damping any reflections in his immediate vicinity, while still allowing him to see the other musicians.



By setting up the singer in the corner of the room, the drapes helped to reduce the levels of spill from the acoustic guitar. The hand-held condenser mic that had been used for the vocals was also changed for the band's large-diaphragm condenser mic.

Etienne had been using his live, hand-held Audio-Technica AE5400 capacitor mic for recording until now, and although this is a good mic we persuaded him to

use the Audio-Technica AT3035 large-diaphragm condenser mic on a stand instead, although this would need a pop shield, which Etienne agreed to buy as soon as possible. He was used to holding the AE5400 mic and using it very close up, but we explained that this caused handling-noise problems, as well as being more prone to wind blasting, while variations in distance could compromise the tone due to the proximity bass boost. It may be a good technique live, but not in the studio! Maintaining a distance of four to six inches from the AT3035 on a stand, with the pop shield between the mic and singer, would produce better and more consistent results, and testing this (without the pop shield as we didn't have one) confirmed that we could get a good basic vocal sound with minimal spill.

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Acoustic & Electric Guitar Tactics

Tackling Nigel's acoustic guitar problem revealed that he didn't necessarily want a natural miked sound, as his rhythm playing also provided the bass end of the mix — the band has no bass player. He'd been using quite a lot of compression to get a warm, sustained low end, but this ended up sounding a little boxy and boomy. As luck would have it, I'd brought along my Line 6 Pod XT, so I suggested we try using that on the Piezo Acoustic setting using its own compressor.

Straightaway this produced a less woolly sound that still had the strength and sustain Nigel wanted, but rather than suggesting he buy a Pod XT (which is really designed for electric guitars), I recommended he try some dedicated acoustic preamps such as those made by Yamaha and Boss, as I felt these would give him more

flexibility. Most such devices also include a sweepable notch filter to help counter feedback, which could be useful in live performance. I also mentioned to Nigel that I'd seen some acoustic guitar feedback stoppers that were effectively large rubber bungs that covered the sound hole. While I haven't tried these personally, they seemed worth checking out for live use, as they don't require any modification to the instrument and they are relatively cheap.

One possible method of using a DI processor live is to feed one of the stereo outs to the combo for on-stage monitoring and the other to the PA for the main front-of-house sound. I often do a similar thing with my Pod XT for electric guitar, and at smaller venues this works extremely well. Where the processor has

Diagrams: Tom Flint



To improve the vocal recordings still further, Paul recommended adding a pop shield to the vocal setup, for a start. In addition, he suggested hanging a duvet from a rail to enclose the back and sides of the singing position, helping to reduce spill while keeping sight lines open.

balanced outputs and is being connected to a device with balanced inputs (such as a mixer), balanced leads should be used to cut down on interference and hum.

Simon plays a really nice Japanese Fender Squire Stratocaster, and while his Marshall sounded fine as a live amp, the DI'd sound was pretty lacklustre, with no bite or energy. Again we tried my Pod XT as an alternative, this time using a mildly overdriven patch augmented by a little reverb and stereo echo that I'd set up for my own use. Simon liked this much better, but before putting a Pod XT on his shopping list, he produced a Korg AX1000G guitar processor from under the bench, so we set about re-programming it to see if we could get a similar sound. The closest we got was using the Modern High Gain amp setting, but when switching back to the Pod XT, we always preferred its sound, which was less fizzy and more focussed, as well as having a better sense of existing in a real space. So I think he might be going shopping in the very near future...

Simon was also experiencing the inevitable single-coil pickup hum problems that plague Strat players. There was one fluorescent light in the studio that could be turned off, but the proximity of so many power amplifiers and other devices with transformers and wall warts, combined with the relatively small size of the studio, meant he couldn't really move far enough away from all the potential sources of interference to avoid the hum altogether. Being a Strat player myself, I suggested he try Kinman replacement pickups, as I've found these to be excellent both for recording and live performance. These are humbuckers designed to look and sound like the single-coil pickups they are designed to replace, and they are very effective in reducing hum without compromising tone. Simon, being from New Zealand, was wary of anything Australian, but seemed prepared to check them out anyway!

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Tweaking The V-Drums

The Roland V-Drums were the least troublesome of the sound sources, though I was told the snare drum was re-triggering sometimes when adjacent drums were hit. On checking through the menus, I found that no crosstalk cancellation had been dialled in for the snare drum, so I adjusted this and the problem went away.



Another strategy for reducing spill in the room while recording was to use a modelling guitar preamp for recording the electric guitar, allowing a floor wedge to be used at low volumes for monitoring purposes. Paul's own Line 6 Pod XT (left) was able to create a sound the guitarist was happy to record with, and because the band's own Korg AX1000G (right) was unable to do so, despite concerted programming efforts, they decided to budget for a new modelling preamp.

The band also felt the kick drum wasn't coming over strongly enough in their mixes, but this was a simple matter of choosing a kick with more attack to it, and the TD8 'brain' module has lots of kicks to choose from. A little rebalancing of the drums sorted out a usable, basic sound, after which I set up a shorter, bright reverb using the 'Bathroom' setting with 'Glass' surfaces.

Nigel had been adding compression to the drum kit via the Tascam's effects section, and the fast attack time he'd been using had robbed the drum sound of some of its impact. My feeling was that, as the V-Drum sound was fully produced right out of the box, it shouldn't need any compression at all and only minimal EQ — possibly a bit of top lift to liven up the cymbals and snare. So, we bypassed the compressor in the Tascam 788 and immediately improved the drum sound.

Nigel also asked about adding overall reverb to the drum sound while mixing, but we advised against this on the grounds that kick drums should be left relatively free of reverb and the TD8 can provide all the reverb needed on the other individual drums and cymbals. Although track limitations in the Tascam 788 might mean recording the drums in mono sometimes, this shouldn't be too detrimental to the overall sound.

While on the subject of compressors, we also suggested not compressing the DI outputs from guitar processor boxes such as the Pod XT, as these devices include all the processing needed. Where such devices have to be recorded in mono to save tracks, a little stereo reverb may be added while mixing to restore a sense of space, but that's pretty much all the processing that should be needed. We set up a nice generic reverb on the Tascam 788 that was useful over the 1.8-2.2s range, and that worked nicely on vocals and other sounds that needed it. I then fine-tuned the vocal compressor setting using a 4:1 ratio, a fast attack and a release time of around 200-250ms.

Sadly, the Tascam compressor doesn't have a gain reduction meter, which is a serious omission on a device designed to be used by musicians who may not have a lot of signal processing experience, so the threshold level had to be set entirely by ear while listening out for the artefacts of over-compression. It isn't possible to create a compression preset where the threshold doesn't need adjustment, because the correct threshold setting depends on the recording level of the track and also on the dynamics of the recorded signal. Our advice was to lower the threshold until the vocal level sat evenly in the track, and not to take it any lower than necessary.



The sound that the band were getting from their Roland V-Drum kit wasn't working in their mixes, so Paul set about choosing different sounds and balancing them more suitably. He also adjusted the effects settings to avoid having to tie up the multitracker's effects for the task.

As I'd only taken one Pod XT with me, and because we didn't have a multiple-output headphone amp (or multiple sets of headphones come to that), we set up a test recording using minimal speaker monitoring levels, where Simon used my Pod XT for the electric guitar and Nigel DI'd from his Carlsbro combo as usual, but with a jack inserted into its phones socket to kill the speaker output. Etienne needed a little vocal monitoring from the floor wedge, but we were still able to make a recording with very little spill. We were confident that with a suitable headphone system in place, the end result would have been even better. The acoustic guitar didn't sound too good, but using a suitable acoustic processor instead of the amp would fix this. Simon was happy with his guitar sound using the Pod XT, and the vocals came over clear and natural sounding, needing only a little compression and reverb to make them sit nicely in the mix.

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The Band's Comments

"As novices in the recording business, our recordings to date have sounded very amateur, but in an evening we have made a 'quantum leap' thanks to Studio SOS. Our recordings are much cleaner and less muddled and somehow 'smaller' but more powerful, giving much better separation between the instruments. We just need to fine-tune our understanding of reverb and compression and experiment with our new electronic toys — we now have a Pod for Simon's guitar and a Yamaha AG Stomp for Nigel's acoustic, and Etienne is thoroughly enjoying a 'hands free' mic!



"We have also just got a five-way headphone monitoring system, and we are now confident that we can achieve good 'live' takes with only backing vocals being overdubbed. We have even bought a bass for some temporary overdubbing until we find a bassist or fathom out how to get a drone out of the Roland's kick drum. Our biggest concern was that our equipment and our studio were never going to give us the results we needed, and we are extremely grateful to Paul and Hugh for re-educating us in this respect. Its amazing how simple things become when you know how to do them! Thanks very much to Paul and Hugh for a whole heap of invaluable advice. Almost all of your suggestions have been put into place already, and the rest will soon follow — and what a difference they have made!"

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Replacing The Bass

During the session, the band were also discussing what to do about their lack of

the extent of further processing
required. **SOS**

Published in SOS March 2004



Own up — who's scoffed all the biccies?

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Studio SOS

Chris Brockis

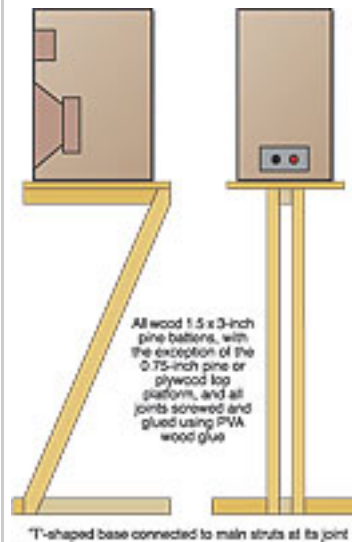
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People : Studio SOS

Chris Brockis had been having trouble mixing in his attic home studio, so he enlisted the help of the trusty SOS team.

Paul White



For a start, Paul suggested mounting the monitors on sturdy stands, rather than on the desktop, in order to improve their accuracy, so he sketched out a suitable design for Chris then and there, as shown in the diagram to the right.

When most people tell you they built their own studio, they usually mean they lined their garage with wood battens, rockwool and gypsum plasterboard, but Chris Brockis personally built a complete extension to his house, which left him with a reasonably large loft space to use for his studio. The studio is accessed via a spiral staircase from the new kitchen area of his Weybridge home, but at the time of our visit it had not been acoustically treated in any way. As the kitchen contained chocolate biscuits, it was a few minutes before we finally ascended the

stairs...

Most of the work Chris does is in collaboration with other musicians, and he's currently working with Francis Firebrace, a charismatic story-teller/songwriter from Australia, and Mick Rogers, the guitar player with Manfred Mann's Earthband. Mick's own material, which features not only his distinctive guitar playing but also some excellent vocals, normally starts out in Mick's own home studio and is then transferred to Chris' *Cubase SX* setup for completion and mixing. For Mick's forthcoming album, they wanted to be able to record direct to *Cubase* in order to improve the creative process and the overall quality of the finished product. Chris realised that the existing environment needed to be improved, particularly in terms of monitoring and layout, but on a limited budget the improvements would need to be prioritised. So Chris called SOS for help.

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The Attic Studio

The studio room is rectangular, but with a flat-topped apex roof (50-degree pitch) coming to within around 1.2 metres of the floor on either side. Chris had his monitors set up at one end of the room, where the wall is a little under 3.5m wide and the full length of the room behind him is 5.8m, with the spiral staircase emerging close to the rear wall. Given the dimensions of the room, this orientation for monitoring seemed the best bet. Chris was mixing on a fairly old pair of KEF Concord hi-fi speakers, which were perched on fairly wobbly stands that placed the tweeter below the ideal ear height — presumably these were designed for 'easy chair' listening. They were also positioned at each end of his equipment desk, and angled to focus a metre or so behind Chris' chair, so he wasn't getting a very accurate sound picture. A pair of JBL Control 1s were wired in as a secondary reference, but Chris didn't have a speaker switcher and so had to change cables over every time he wanted to change monitors. He had also wound in a huge amount of bass boost when using the JBLs to try to compensate for their inherently shy bass response, with the expected result of a very muddy and tuneless bass end.

A cursory glance around the recording hardware showed that much of Chris' gear had been accumulated during the '80s, as there was an old Yamaha SPX90, an Aphex Type C Aural Exciter, a Yamaha FX800 multi-effects box, a Yamaha GC2020 compressor and a Korg M1R rackmount synth. The biggest chunk of hardware by far was a Tascam 388 combined open-reel eight-track recorder and mixer, which Chris was now using mainly as a mixer. Aside from the 388 mixer, very little of this gear was now used, as Chris tended to record, manipulate and mix everything inside *Cubase SX* (including adding effects and reverb), which was running on an extremely quiet Syrinx PC put together for him by Andertons Music. The audio interface was an Edirol DA2496.

Chris was keen to optimise his work space, so Hugh suggested that the 388 recorder/mixer be moved to the left-hand wall, which would give Chris more room for his computer equipment and, more importantly, allow the monitors to be

positioned more appropriately. His master keyboard could then be positioned immediately to his right, so that he could get to it simply by swivelling his chair. However, before changing anything, we wanted to get an idea of how his existing monitoring system sounded, so we played some of his mixes and compared them with a Sting CD played through the same system.

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Hi-fi Monitoring Problems

Although not terrible, the KEFs sounded somewhat dull and unfocused and had no real depth of bass, which meant Chris was using EQ to compensate for the speaker deficiencies, more than to correct his mixes. Chris had already told us beforehand that he was using hi-fi monitors, so I brought along the Mackie HR624s that normally provide the rear surround monitoring in my own studio to see if they sounded any better. Chris said he'd already tried these Mackie speakers in a music store and thought they didn't sound particularly great, but as music stores are not the best place to audition speakers, he agreed to give them a try.

In the absence of stands, we fixed them to the tops of the KEFs using blobs of Blu-Tac, then moved the KEFs further back and angled them in to establish a more traditional 'monitoring triangle'. On replaying the same tracks, the Sting CD sounded much more crisp and precise, with a nice tight, tuneful bass end. On the other hand, Chris's mixes sounded rather bass heavy, confirming our suspicion that he'd been misled by his monitors. Chris agreed that the Mackie monitors sounded great in his room, so we left them set up for the rest of our visit and Chris soon added them to his monitor short list.

We also suggested some hefty speaker stands around one metre high, which could either be commercial hi-fi stands, ideally filled with sand, or (as he was obviously handy with tools) he could make a pair of wooden stands similar to those I use in my own studio — I discussed this design in detail back in SOS November 1995.

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Acoustic Treatment

Considering that the room had no treatment other than a carpeted floor, it didn't sound bad, but there were some early reflections clouding the stereo imaging, so we decided to try some minimal acoustic foam treatment. Acoustic foam isn't a complete solution, otherwise the top pro studios would use nothing else, but it is useful for absorbing the mid-range and high frequencies that make early reflections so distracting. However, it has to be used in moderation, as covering too much of the room in foam will damp out all the high end leaving the bass and lower mid-range to run riot — a certain recipe for a boxy-sounding room. The system we usually follow is the tried and tested one of placing around half a square metre of foam on the side walls at the engineer's head height, extending

towards the speakers from just behind the engineer's head. If you were to place a mirror on the side wall in such a position as to make

the speaker visible from your monitoring position, that would be where you'd put the centre of your foam panel.

It can also help to do the same with the ceiling, but because of the sloping angles in Chris's room, reflections weren't such a serious problem, so we decide to try it without. However, we did feel that a further panel in the middle of the front wall, between the monitors, would help a little, as otherwise there was a risk of getting 'secondhand' reflection from the flat hard surface, a little over a metre in front of the mixing position. And, of course, no Studio SOS would be complete without a duvet, so to tame the back wall a little, we suggested the simple ploy of fitting a curtain pole and then draping a king-size duvet over it. Yes, this is a pragmatic approach rather than a calculated solution, but when you're on a tight budget, a little low-tech treatment can make a significant improvement to an otherwise indifferent room. We also suggested draping a duvet behind the vocalist when doing vocal recording, so Chris, being ever resourceful, said he'd build a suitable frame out of copper plumbing pipe and use that to hold up the duvet.

Because Chris' room looked so nice and new, we didn't want to do anything too permanent to it, so instead of sticking Auralex foam panels directly to the wall, we wedged them roughly in place just to test the result, then suggested to Chris that he fix them to MDF boards and then fix those to the wall using hooks. That way, if he decided to rearrange the studio at some future date, he wouldn't be left with patches of adhesive on his paintwork.



Temporarily placing a panel of acoustic foam behind the monitors facing the monitoring position confirmed that this would improve the sound by avoiding strong reflections from the wall.

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Optimising *Auto-Tune*

Once we had the monitoring behaving adequately well, Chris asked us to advise on some issues relating to his mixes. His first question concerned the use of *Antares Auto-Tune*, where he was using it to pitch-correct some layered vocal harmony parts. He'd set up the correct scale, but on some of the parts the pitch-correction was too fast, resulting in that familiar quantised, vocoder-like sound. He'd also used the vibrato feature of *Auto-Tune* to add a little artificial vibrato after a short delay.

We managed to improve things by setting the correction speed control to around the halfway point, which still held the tuning, but without taking away all the natural inflections of the voice. What's more, as the natural character of the voice

was better able to get through, the addition of artificial vibrato became less necessary, so we were able to reduce the vibrato depth on some of the parts. We also staggered the vibrato delay times on the different plug-ins to create a less regimented effect, and varying the vibrato rates slightly is a good idea too. As a rule, always use as little *Auto-Tune* as you can get away with if you want the most natural-sounding vocals, though you can generally afford to use it a little more heavily on backing vocals than on lead vocals.

The other mix issues concerned EQ and dynamics, two common areas of concern. When working with Mick Rogers, Chris usually took a stereo mix of a part-finished track from Mick and then added further parts, but he'd also been adding a lot of EQ to the original stereo track, partly to compensate for his monitors. Once we had the monitoring sorted out, we needed less drastic EQ, and in some cases Chris had been using large amounts of EQ with quite narrow bandwidths, which almost always sounds unnatural.



Here you can see the mix processing settings used to polish Chris's mixes.

Instead, we created a gentle smile curve by boosting at 80Hz over a wider range to bring out the bass drum and bass parts, cutting the lower mid-range at 180Hz and then adding in some high-end sizzle at 12-14kHz with a wide (low-Q) setting.

The amount of cut and boost we had to add using the *Cubase SX Channel EQ* plug-in was rather more than you'd expect if you were using an analogue equaliser and, as every digital equaliser behaves slightly differently in this respect, it's impossible to give a recommended amount for other systems. However, the screenshot shows what we ended up with on *Cubase SX*. The result of applying this smile curve was to add a little weight to the low end while clarifying the mid-range and adding more definition to the high frequencies. Despite the apparently drastic-looking EQ curve, the amount of change when bypassing the effect was still quite subtle.

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Cathode Ray Trouble

We had noticed that handling *Cubase SX* on a single 19-inch CRT monitor was a bit of a pain — we were forever opening and closing windows, so we suggested that a double-headed graphics card and two 15-inch or 17-inch LCD screens would be more useful, as he would be able to leave multiple windows open and, as a bonus, he could record electric guitar without the monitors causing buzz — a particular problem, as Mick plays a guitar fitted with single-coil pickups. The CRT was also very bulky and was potentially causing imaging problems because of reflections. LCD monitors are far less intrusive in this respect.

Cubase SX Dynamics Tips

Chris also wanted to know how best to apply dynamics processing to the final mix. I first tried the *Cubase SX* multi-band compressor, but I could find few of the parameters I was expecting. There seemed to be an overall graphical control that affected the amount of compression in all three bands at once, and it was easy to change the crossover frequencies and the gain in each of the three bands, but I couldn't find the rest of the parameters I was familiar with, so I used the full-band compressor in the dynamics section instead. To increase the density without squashing the track, I set a low threshold and a ratio of just 1.2:1, then fine-tuned the threshold to give in the order of 6dB of gain reduction on the signal peaks. This particular processor has an automatic release function, which seemed to work fairly well so I left it on, and I set the attack time fairly fast. If you find that transient percussive sounds are losing their definition, you can increase the attack time to 20ms or so, but with such a low ratio, we had no problems of this kind.

Having tried these overall fixes, we identified some balance and EQ problems in the original stereo mixes that could be better resolved at the mixing stage, and suggested that Mick brought his mixes over as separate tracks rather than as a simple stereo submix. Chris agreed that this is how they intended to work for Mick's forthcoming album. They'd already ended up resorting to strategies such as layering on another kick-drum part because the one in the stereo mix lacked power or definition. Being able to fix or replace the original part prior to mixing would be much more satisfactory.

There were also some arrangement issues, especially on a track Mick had written called 'U Don't Love Me' where his vocals sounded uncannily like Peter Gabriel (the track featured a distinctive drone part created by detuning the guitar by four semitones and then processing it via distortion and a wah-wah pedal). This was a great song, well played and well sung, but the drum part featured a wall-to-wall splashy ride cymbal that stole all the space from the mix. This could have been replaced with a duller or more percussive type of cymbal like the Zildjian Earth Ride, or even omitted altogether.

Chris had also tried using the *Cubase* time-stretch facilities to fix the tempo of a slightly slow mix that Mick had brought over, but even though the amount of tempo increase was modest I could still hear the time-stretching artefacts, which had the effect of clouding the mix and making it sound messy — something that was much more evident when heard over the Mackie monitors. It should have been possible to do a better job using a more sophisticated third-party plug-in to do the time-stretching, but where you have the time to start over and get the tempo right, I feel it's very important to do so, especially when you're trying to process a significant chunk of the mix and not just the odd track.

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Chris's Comments

"It was great to meet Paul and Hugh. Their expertise has helped me to create a much improved studio environment relatively cheaply, barring the expense of new monitors, of course! When I first set up the room in 2002 I knew that improvements would be required, but I didn't know where to start and where the major gains were to be had.

"After a cup of tea (and some chocolate biscuits, of course!), Paul plugged in his Mackie HR624s. I was amazed at the improvement over my KEFs. Then, with the Auralex strategically placed, I was astounded by the improved clarity of the soundscape. I had not expected such a major improvement from just a couple of panels! When Paul and Hugh left I had a clear idea of what needed to be done. I had just two weeks to complete everything for their next visit during the Sounds Expo show.

"The biggest issue for me was monitor selection. Before the visit, I auditioned most of the current crop of monitors, including the HR624s, at my local music shop. At the end of the session I was none the wiser as to what would be appropriate in my studio and I was reluctant to part with the readies. I was also surprised at how the same speakers could sound so different with music of different genres. One particular pair sounded clear and articulate with simple pop and R&B, but couldn't handle more grungy rock music at all. On the basis that Paul's Mackies sounded great *in situ* and handled all genres well, I decided to go with them. I had intended to build Paul's design for wooden stands, but finally went for sand-filled Atacama Nexus 10s.


"The shape of the roof made designing the suggested 'duvet vocal booth' screen a bit awkward. My wife Caroline's inspired suggestion was to try the chrome clothes rail in our spare bedroom — it's adjustable in height, collapsible, and only cost £6.40 from Argos. However, I found that it was not quite tall enough, even at its maximum extension. As luck would have it, though, the upper vertical tubes can be replaced cheaply by longer lengths of 22mm standard copper pipe. I bought two clothes rails and some copper pipe for less than £25! The rails are linked at the top for stability and I have put 45-degree bends in the horizontal pipework to make the screen semicircular. I also bought a 13-tog king-size duvet, with a cover to match the Auralex foam, so it all fits together rather well.

"The equipment has been moved around as suggested by Hugh. The PC workstation surface has been extended with an offcut of 38mm worktop, which cost £6. The original BT patchbay has gone in favour of a Behringer 48-way TRS patchbay. I have added a separate workstation for my Atari sequencer and there is plenty of space for Mick's 24-track digital recorder and outboard gear. The overall improvement is nothing less than amazing, both acoustically and ergonomically. The improvised recording booth has noticeably improved both vocal and acoustic guitar recording. Mixing and monitoring is now a delight. Another job well done guys!

"Incidentally, Mick has been singing for as long as Peter Gabriel, and has appeared on about 20 albums, so I sometimes think that Peter Gabriel sounds uncannily like Mick Rogers, rather than the other way around..."

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All's Well That Ends Well

Chris agreed to sort out some new monitors and stands as well as rearranging the gear and fixing the foam tiles so that we could visit him during the forthcoming Sounds Expo show in London to photograph the finished job — find out how things turned out in the 'Chris's Comments' box. We got a feeling of *déjà vu* as we left, because this time last year we left a Studio SOS visit in South London and drove straight into the worst blizzard of the year. This time we managed to get as far as the M40 before the rain turned to snow, and before long we were crawling home at 10mph with cars and lorries slipping and sliding around us. Such fun! 

Published in SOS April 2004

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Studio SOS

The Arcades

Published in SOS June 2004

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People : Studio SOS

This month, the SOS team help The Arcades to rock even harder than before!

Paul White

We were lured to The Arcades' studio by the promise of chocolate Hobnobs and French Fancies, the mission brief being to improve the way they recorded their four-piece rock band. Their influences, at least sound-wise, were Led Zeppelin, Deep Purple, and AC/DC, so at least we knew what we were aiming for. When we arrived, we discovered that the studio actually comprised two wooden outbuildings which had been built by brothers Greg (drums) and Graham (lead guitar) who both live where the studio is located. One room was for use as a control room and the larger one was the live room. Greg had completely wired the studio with balanced wiring for the best signal path from the live room to the control room — they were separated by a couple of metres and, as the band do their own engineering, it was pretty much a matter of 'last one out hit record'!

The live room was almost entirely lined with mattresses, foam, and reclaimed sound tiles, which in combination with the bass-trapping nature of the wooden building gave a reasonably dead but fairly well-balanced sound, not unlike that of many professional studios during the '70s. There was some sound leakage to the outside, but as the studio is out in the countryside this wasn't a problem.

The control room measured about 2.5 x 3.5m and was lined out with batten and plasterboard. Some acoustic foam tiles were fixed to the rear wall, ceiling, and side wall, but we noticed straightaway that there was a patch of bare wall to the right of the mixing position that was likely to throw back reflections from the monitors, so we suggested adding another foam tile or moving one of the existing ones. Similarly, although there were two foam squares on the ceiling, the part most likely to produce reflections hadn't been covered, so we suggested doing the same there.

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Monitoring Assessments

As has now become routine on our visits, the first thing we looked at was the monitoring system, which comprised a pair of Genelec 2029As and the matching 7050A subwoofer. The speakers were set up at the correct heights and angles on rigid shelves either side of the band's two computer monitor screens. On playing back some commercial material, we felt that the sub was turned up a little too high, and set about rebalancing it by ear. Because of the natural bass trapping of the building and the fortuitous location of the subwoofer under the mixer table, the bass end was reasonably even (no noticeable level differences between different bass notes) so no further tweaking was necessary once the level had been turned down by about 2-3dB. However, it is difficult to set up subs accurately without specialist test equipment, so it is recommended that mixes be played on as many different systems as possible and that then, depending on the results, the subwoofer level be adjusted again accordingly. If the mixes are generally perceived as being bass light, then the subwoofer level is probably too high, which causes the engineer to reign in the bass end to compensate. Naturally, the exact opposite is the case if the subwoofer level is too low, as that fools the engineer into adding more bass.



The hole in the front of the kick drum's front head was too small to allow a suitable mic position, so it was removed. The sound was then damped to taste using a pillow resting in the drum shell, and some extra beater definition was added by taping Paul's RAC card to the point where the beater hit the drum head. Finally, the drum was miked using an AKG D112. This was placed centrally to start with, but auditioning showed that a position slightly to one side produced a better sound.

The Arcades' recording system is based around a PC running the latest version of Steinberg's *Cubase SX*, which they're still getting used to. The audio interface is a MOTU 828 MkII and the mixing console is a recently acquired second-hand Spirit Studio model, though it seems to have a couple of temperamental channels. We didn't have time to remedy this, but Hugh suggested wiring up a TRS jack plug with the tip and sleeve connected together to push into the insert points of the suspect channels, since it appeared that dirty or damaged insert sockets were causing some of the problem. If the insert socket was at fault, leaving the jack in place should fix the problem.

There was also a small amount of outboard equipment, though most processing tended to be done in software. By utilising all the 828's analogue inputs, the band could record up to 10 channels simultaneously, which was more than enough, as only nine feeds were needed to record the two guitars, the bass, and the six drum mics. The band's own Lexicon MPX100 reverb was broken, so I brought along my own Lexicon MPX550, which includes some useful ambience and stone-hallway algorithms that I thought might help us in getting a big drum sound. This was patched into one of the aux sends on the Spirit desk and brought back via a pair of return channels before we started work.

Before trying anything new, we listened to the multitrack playback of a song the band had been working on. It didn't sound at all bad, but it lacked the sense of space and power they were after, and the boys were particularly keen to improve the drum sound if possible. The band were also experiencing spill problems when they all played together, but as is so often the case with this type of music, the recording loses its energy if you try to layer it up as individual overdubs.

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Setting Up For A Big Drum Sound

We decided to tackle the drum kit first, and though the band had a decent selection of mics, they had already told me they didn't have two identical capacitor mics to use as overheads (other than a pair of AKG C1000s, which didn't sound particularly good as overheads), so I brought along my own SE Electronics SE1s, as I thought they'd do the job and they were in a price range the band could afford. These are small-diaphragm, cardioid-pattern capacitor models and are surprisingly good all-round instrument mics for anyone on a tight budget, though pretty much any of the current crop of budget cardioid capacitor mics (small or large diaphragm) from the Far East would do the job well enough.

Before setting up the mics, I checked over the drum kit and found that the kick's front head had only a small hole cut into it, which meant I couldn't position the mic correctly. Greg removed the head of the drum while I checked the rest of his DW kit. He'd tuned the toms using a specialist skin-tension tuning device, which I'd never come across before, so the tension was pretty even and the tone was OK.

However, to get more of a rock sound I tried slackening just one lug on each tom by around half a turn to get a hint of pitch drop after the drum was struck. This worked nicely, but we had too much ring on the toms, so I taped a small wad of tissue, around one inch square, to both the top and bottom heads. This dried up the sound without killing all the natural ring.



The snare was a perfectly tuned, wooden-shelled job around four inches deep, so it had a bright lively tone. It was never going to sound deep and fat, but it sounded good and wasn't too far away from the AC/DC snare sound the band liked. That left the kick, which was again already perfectly tuned, so once the front head had been removed, we adjusted the integral damping pillow and I taped my plastic RAC membership card to the head where the beater hits to get a bit more of a snap into the sound. I did remember to retrieve it before we drove home, through!

For miking, we used the band's own Shure SM57s on the snare and toms, all in the traditional positions around two inches up and two inches in from the rim. Where possible, these were aimed away from the cymbals to minimise spill. The SE1s went up as spaced overheads about eighteen inches above the cymbals, and an AKG D112 was set up on a short boom stand so that it sat roughly in the centre of the kick drum shell, aimed at the beater. The ceiling above the drum kit had been covered with proper, reclaimed acoustic absorber units which soaked up a lot of the cymbal noise and prevented it from splashing all around the room. A test recording showed that the kick drum still sounded a little tubby, so I moved the mic slightly to one side and added EQ boost around those old standby frequencies of 70Hz for depth and 4kHz for click. This sounded much more solid, and when a software gate was applied in *Cubase* to cut out the spill from the other drums it sounded very sweet indeed.



Although the toms were well tuned, they were ringing a little too much, but it was nothing that a bit of masking tape and tissue paper couldn't deal with!

The overheads were picking up a good overall kit sound and bringing up the cymbals, but I rolled some low end off these mics, as I find that often gives a clearer sound overall by avoiding conflict with the close mics. Where the individual drums are going to be panned to create a stereo image, care must be taken to get the close-mic pan positions to match the image captured by the overheads, but in most cases you can get a very solid sound by leaving the close mics panned centre and then relying on the overheads and stereo reverb to add a sense of width and space.

I've always had problems when miking double-headed toms, in that they ring almost continuously, because they invariably resonate in sympathy with the kick drum. There's little you can do about this other than damping all the life out of them, so my preferred option is to gate them after recording, which is exactly what we did. To avoid producing an unnatural and completely dead sound, we set the gate range to about 12dB, so that it never actually closed completely, just reduced the spill by the 12dB. The attack was set very fast, with a short hold time and a release time set to follow the natural decay time of each tom. This tidied up the sound immeasurably, but if you have the patience it's even better to go

through the tom tracks manually and silence all the sections between tom hits using your software's waveform editor. This doesn't take as long as you might imagine, and it ensures that nothing comes through on the tom tracks except toms. It also gives you the opportunity to do a destructive gain change on any hits that are too loud or too soft. The other advantage is that this way you can ensure the initial transients are retained in all their glory, whereas some analogue gates tend to remove or significantly reduce that initial transient 'thwack'. If you're using a software gate, engaging the lookahead facility can avoid the transients being damaged, as the gate will open slightly before the transient arrives.



Paul experimented with adding different patches from his Lexicon MPX550 reverb unit, and this yielded a bigger, more spacious drum sound.

Other than the kick-drum EQ and the low-end roll-off on the overheads, very little EQ was needed to get a nice punchy drum sound, after which I tried adding a Lexicon ambience program to the whole kit to try to approximate that 'recorded in a rock star's mansion' sound. This did actually work quite well, and while adding reverb to kick drums isn't usually a good idea, short ambience programs work fine, adding depth and space without muddying the sound. As an alternative, I also demonstrated the Marble Foyer patch in the MPX550, which gave a bigger, more live sound that was still not too busy. Of course nothing sounds quite like a real live room, apart from perhaps a convolution reverb with a suitable impulse response, but we were all pretty happy with the overall effect.

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Adding The Guitars & Bass

Having got the drums sorted, it was time to turn our attention to the guitars and bass. The bass had been DI'd from a socket on the amp, and though this doesn't always sound great it was a good starting point. However, the bass and guitars had been tuned down a semitone for artistic reasons (apparently it sounds heavier!), which had left the bass with a noticeable amount of fret rattle that was plainly audible on our recordings. Rolling off some of the high end helped, but the only true solution was to have the bass set up professionally to get rid of the rattle at source. For the sake of our tests, we plugged a jack into the bass



amp's extension speaker socket to kill the speaker feed (this is OK on solid-state amps but not wise on tube amps!) and everyone monitored on headphones while playing.

A liberal helping of the Waves *Renaissance Compressor* plug-in helped even out and thicken the bass sound, and, other than the odd rattle, it sounded quite acceptable with very little in the way of EQ. However, I do feel that you get a much punchier rock bass sound from modelling preamps such as the Line 6 Bass Pod or Behringer Bass V-Amp than from a straight DI, as DI'ing loses the important coloration of the speaker cab. Even the guitar versions of these modelling processors produce great bass sounds if you team up something like a Fender Bassman amp model with a 15-inch speaker model.



Turning the guitar cabinet round to face the absorbent acoustic treatment on the studio wall significantly reduced the acoustic spill into the drum mics, and a small gap left between the speaker and the wall allowed an AKG C1000 to be positioned for a very respectable recorded sound.

We still had to deal with the guitar spill problem, and I also felt the guitar sounds weren't focused enough, so I spent some time adjusting the amp controls while the guys played. Fortunately, they both used power soaks (THD Hot Plates from America) so that they could play with their big Marshall amps (a 1959 SLP and a JCM800) set to '11' without deafening each other — though I did notice a few packets of disposable ear plugs in the live room! By turning off the Bright switch on the power soaks and restoring the bite using the amps' Presence controls, I found that we got a sweeter, less gritty tone that better suited the musical style they were aiming for.

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Kill That Spill

Sorting out the spill problem was easy — we simply turned the sealed-back 4x12 cabs around to face the mattress-lined walls, leaving just enough space to get a mic stand's boom arm in. Not only did the wall treatment massively reduce the sound from the guitar amp bouncing back into the room, but it also minimised the amount of drum sound reaching the guitar mics. While SM57s are an obvious choice for recording guitar cabinets, it's worth trying every mic you have, as some combinations can produce surprising results. We decided to try the AKG C1000s, as they have quite a smooth top end not unlike that of a dynamic mic, and these were placed up close and personal, aimed directly at the centre of one of the speaker cones. This is just a starting position, and if the tone is too hard you can warm it up by moving the mic slightly towards the edge of the speaker cone. You can't get a great guitar sound just by following textbooks — you really do have to experiment with mic types and positions to get the best results in any given situation.

At this point we made a test recording and found that the amount of spill between guitars and drums was minimal. The guitar spill to the drum overhead mics was around 25dB down, and likewise the drums reaching the guitar mics. This separation provided far more clarity to the sound overall, and allowed greater scope for individual processing of tracks without spill colouring the effects. It also made it practical to compress the guitars without pulling up loads of drum spill, and to put ambience reverb on the drums without making the guitars sound like they were in a bathroom.

The sound we got was already an improvement on what the band had recorded earlier, though I felt the guitars needed more bite, but without making them sound gritty or edgy. Even though the two guitarists had very different guitars (a Fender Stratocaster and a Gibson Les Paul), the solution was the same: 2-3dB boost at around 700Hz, which gave the Les Paul more of that AC/DC Gibson SG honk and made the Strat sound less brittle. When I first tried these EQ settings on the soloed guitar tracks, the band were a little unsure, but once the rest of the mix was up and running, it was agreed that the guitars sounded more punchy and sat better in the mix.

Another small tweak we made was to reduce the amount of overdrive used on the lead guitar part and then to insert another *Renaissance Compressor* to maintain the sustain. Again the result was more clarity and punch. If you record using too much overdrive, you run the risk of ending up with a very indistinct 'snails in a blender' sound that eats up all the space in the mix but still doesn't cut through. As an experiment, we also inserted the band's stereo SPL Vitalizer into the stereo drum submix and used it to add more weight to the low end. The high-end controls were then tweaked to add a little more shimmer to the cymbals. Provided that these devices are used with care, they can add weight, clarity, and depth to a track, submix, or full mix, but the secret is not to overuse them.



Because both guitarists were using power soaks (inset), monitoring levels in the recording room could be kept fairly low. It also meant that Paul could comfortably stand by the cabs tweaking the amp controls while the guitarists played, even though the amps were cranked up all the way, so he could quickly optimise the recorded sound at source.

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Mastering Tweaks

By the end of the session, we had recorded a complete backing track that only

needed the vocals to be overdubbed, and the band were all surprised by the difference some simple adjustments and techniques had made to their sound. The band then asked about mastering processes that could be applied to polish up their finished mixes. The difficulty about advising on this subject is that every track is different and, more importantly, that you can only make accurate judgements if you can rely on your monitoring system to be accurate. Some people also believe it's impossible to make a good-sounding master using plug-in processing, but I dispute this — great tools obviously help, but if you have good ears and good monitors, you can do wonders with relatively simple plug-ins or pieces of outboard equipment.



With the bass guitar DI'd and the guitar cabs turned towards the walls, there was ample separation between the different instruments for the whole band to record together in the same room.

The reason material is mastered at all is largely to do with making all the tracks on an album match each other tonally and in relative level (which must be judged subjectively rather than by meters alone), but these days there's also a demand for more loudness and also for making the music sound even better than the final mix. The main tools are EQ, compression, and limiting, where careful use of a nice-sounding equaliser can be used to tame hot spots, scoop out a boxy mid-range, and add a high-frequency gloss. Most times this ends up being a variation on the classic smile curve, where the mid-range is dipped slightly and the high end (12-14kHz) is boosted by a couple of decibels using a very low-Q setting. However, you can't be too formulaic about this — you have to understand what the mix needs and then use modest amounts of EQ to achieve that. As the band have a Vitalizer, they could use that to do much the same thing.

Compression is also important, but unlike individual track compression, where you typically set a high ratio and a high threshold so that the compressor only stamps on signal peaks, the settings used for mastering tend to involve very low ratios of between 1.1:1 and 1.2:1 using threshold settings of -35dB to -40dB. This means that the whole dynamic range of the music is gently squeezed, and gain-reduction readings of 3-4dB are typical. I usually put the compressor before the equaliser, but others prefer to compress afterwards, so try both to see if you can hear a difference. The final stage is to apply limiting so that you can push up the



Getting the right mix of guitars and drums initially proved tricky, but once the guitars had been EQ'd a little, Hugh was able to find a good balance.

average signal level, and as before a gain reduction of around 4dB on signal peaks is usually enough to achieve the desired result without over-processing the sound.

As the band had a good selection of Waves plug-ins, my approach would be to try the *Renaissance Compressor* first, followed by the *Q4* parametric equaliser, though a version with more bands (up to 10) is available if there are any awkward areas that need individual attention. The last stage would be the *L1* limiter, which allows the user to set the peak level (usually around 0.5dB from clipping) after which the input gain is increased until the desired amount of gain reduction shows on the signal peaks. Getting all this just right requires practice, but if you err on the side of under-processing rather than over-processing, you should be OK. Of course it also helps to compare every stage of your mastering with a commercial record in the same musical style played back over the same monitor system at the same level. In this case the band were after a Led Zeppelin kind of sound, but some of Led Zeppelin's early records sound lacking in punch compared with modern productions, so choosing a good variety of material would be safest.

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Ready To Rock

Despite us only being on site for a few hours, we all worked together effectively to improve both the guitar and drum sounds while also solving the spill problem. Once we'd turned the guitar speaker around, soloing the individual guitar and drum tracks showed virtually no spill other than on the drum overheads, where a very low level of guitar was audible.



Listening to the new final mix side by side with their previous efforts, the band could already hear a big difference. However, comparisons to commercial releases showed that some careful mastering processing would still prove beneficial.

The band had commented that they found it hard to record the vocal overdubs, which they had been doing facing directly into a heavily foam-lined corner of the room. This arrangement meant that the foam was soaking up the voice, making it hard for the singer to hear himself — he consequently ended up straining to sing louder. Furthermore, this setup wasn't preventing the ambient sound and reflections from reaching the mic, which was facing out into the room behind the singer.

Hugh suggested that a better way of working would be to place the singer a little out from the corner, looking out into the room, with the microphone facing into the corner. That way, the vocalist wouldn't feel like a dunce sent to the corner, and would find it easier to sing at a more comfortable level, while the foam absorbers behind him would prevent spill and ambience from being reflected back into the

microphone.

The only issue that needed outside help was the setup of the bass guitar to avoid the string/fret rattle. As mentioned in the main text, I think a bass recording preamp would give a more suitable sound, but the band also need to set up a better monitoring system with a headphone distribution unit that allows them to control their individual monitoring levels. As these are now so cheap, that is easily resolved. **SOS**

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The Band's Comments

Recording as a band gives us the chance to keep the magic that happens in the room, so reducing the spill was essential and Paul and Hugh were able to give us hints and ideas on how to achieve this. We've since bought a headphone amp which is now enabling us to record live without ruining our ears — a perfect solution. The drum sound we got on the day was also good, and the reverb unit certainly gave the kit a sense of power. And what a relief to have someone else in the other room to hit record!

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Published in SOS June 2004

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Studio SOS

Peter May

Published in SOS July 2004

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People : Studio SOS

The SOS team help Peter May to brush up his drum sounds and put more life into his mixes.

Paul White



Although the toms and snare sounded acceptable on the first visit, on his return Paul optimised the sound with some careful tuning and damping.

Peter May's studio occupies a room at the rear end of the ground floor of his house in a space known as 'the void', a room half set into the ground, which certainly helps with the sound isolation. Peter is a school teacher, but was fortunate earlier in his life to get a teacher placement at Abbey Road studios, which gave him the bug for recording and lead him to question his choice of career!

Having worked his way up through TEAC four- and eight-track machines, Peter has been recording since the late '70s, but has now moved over to a computer-based system comprising a Windows machine running Steinberg's *Cubase SX*.

He has also recently invested in a new monitoring system comprising Genelec 1029As with a matching subwoofer. Although at the time of our first visit he still used his Soundcraft Spirit powered mixer for monitoring his computer mixes and for playing back CDs and so on, recording was done through a recently acquired Allen & Heath GS3 analogue console, which feeds his 10-input Terratec audio interface.

The room itself is separated into two areas by a supporting buttress, and all the walls have been covered using thin wood-effect panelling on battens with lightweight rockwool behind. Although this makes the surfaces very reflective, the sheer amount of stuff in the studio provides a lot of dispersion to keep this nicely under control, and the panelling (more by accident than design) seems to act as quite effective bass trapping.

Peter was having trouble getting a good sound from a live drum kit, which he had been recording with just four mics — kick, snare, and two overheads. As he'd recently bought a set of AKG clip-on drum mics, he wanted to explore ways of using these more effectively. He was also having some problems with his mixes sounding lifeless.

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First Things First

The first item on our agenda was to check out the monitoring system. This was not set up symmetrically, nor would it have been practical to do so because of the shape of the room, but we heard a problem with the lower mid-range that wasn't attributable to this. A quick check around the back of the 1029As revealed that Peter hadn't set the DIP switches on the back of the speakers for use with a subwoofer, so the necessary low-end roll-off wasn't happening. Hugh adjusted the DIP switches with the aid of a long thin screwdriver and, once they were correctly orientated, the sound improved noticeably. Playing a track with a busy bass line showed a few small subjective discrepancies in bass level between notes, but they were not nearly as severe as we expected in this room, so we decided to leave the subwoofer where it was (beneath the table supporting the console) and press on.



Peter had been positioning his overhead mics much too close to the kit, so Paul and Hugh moved them higher to get a more cohesive sound. However, the AKG C1000s Peter had selected for the task were rather dull-sounding, so Paul recommended replacing them with a pair of the new generation of inexpensive condenser mics to get a brighter sound.

I offered to tune the Premier drum kit lurking at the other side of the buttress, but Peter rather guiltily confessed that he'd lost the tuning key. Fortunately, the snare

and toms were fairly well tuned anyway and I only needed to apply a small pad of tissue (using masking tape in this instance, though gaffer tape is better) to the upper and lower tom heads to tame the ring. The kick drum was a different matter, as the head was too worn to get any depth in the tone, but I retuned it as best as I could (via the hand-operated tuning lugs), replaced the lightweight pillow inside the drum shell with a heavier sleeping bag (a thick blanket would have been better still), and taped a plastic card to the head to try to get more snap from the felt beater. As a rule, wooden or plastic beaters give better definition to the tone for rock and pop music.

Peter had been setting up the AKG kick mic around a foot in front of the drum, so we repositioned it inside the shell on a boom arm and then experimented with the fine positioning to get the best tone we could. The outcome was a sound with much better definition, but still without any real depth — we weren't getting any depth acoustically either, so a new head has been added to Peter's shopping list.

After raising the AKG C1000 overheads to a more suitable positioning, I played some particularly sloppy drum rhythms so that Hugh could make a test recording on *Cubase SX*. One of the clip-on mics was used for the snare, but we recommended that Peter experiment using his other clip-on mics on the toms after listening to the test recording. The overall sound wasn't bad, but close-miking the toms gives you more control over balance and also gives a more solid sound that's well-suited to rock and pop work.



Peter's original kick mic positioning several inches outside the body of the drum didn't give nearly as much definition to the recorded sound as miking close to the batter head inside the drum.

After recording, we used the gate in *Cubase* to clean up the kick drum, and also applied some EQ. A boost at 80Hz accentuated the (sadly underwhelming) thump of the drum, and a little more boost at around 4kHz emphasised the smack of the beater. A further dip at 150-200Hz stopped the sound becoming boxy.

It is worth noting that drums recorded in small or dead rooms won't sound great unless the right reverb is added, and in many cases the reverb plug-ins that come with most sequencers don't really cut it. What you really need is not a washy or splashy effect, but the sense of being in a real room. The newer convolution reverbs do an excellent job of this, but they are very processor hungry. The best settings to use tend to be ambience programs or responses

taken in real studio rooms, and the host-powered *Cubase SX* reverb wasn't really doing the job. Peter said he was in the market for a better 'in-computer' reverb, and as his studio was almost midway between where Hugh and I live, we agreed to a second visit after he'd made any recommended changes and added some new equipment. I told Peter that I liked the reverbs that came with TC's Powercore, and as the Powercore Element card is now so cheap he said he was seriously considering buying one.

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Vocal & Guitar Miking

Early on we noticed a Rode NT1 mic set up on a boom stand with a pop shield positioned over the end, rather than at the side — the NT1 is a side-entry mic, so the end of the mic is 90 degrees off axis. When we asked Peter about this, he said that he recorded this way because when he had tried singing into the side of the mic it had produced a dull sound. It transpired that he'd been using the *back* of the mic rather than the front, which is perhaps understandable when you've been used to dynamic mics all your life, where there is only one obvious business end. In fact, the Rode mics use a gold stud to denote the correct side to yell into, and as soon as this was explained to Peter, he had no further problems. Nevertheless, as the room had so many reflective surfaces, we did suggest hanging a heavy duvet behind the singer when recording vocals, as this would help soak up any excessive mid-range and high-end liveness.



Paul and Hugh quickly noticed Peter's unusual vocal-miking setup! Given that the NT1 is a side-firing mic, a much clearer sound was immediately obtained by having the singer address the correct side of the mic, as marked with the gold dot.

Peter had also tried using this same mic for recording the acoustic guitar, but with disappointing results. He had felt that perhaps he should be recording in a more dead-sounding area, but I thought the studio room was fine and might even benefit from a sheet of hardboard or MDF on the floor (beneath the guitar) while recording, so as to reflect some sound back up to the mic. We set him up playing his Takamine acoustic guitar and got him to listen to the sound through headphones as I moved the mic around. Peter was surprised at just how much the sound changed with different mic positions and the best sound we achieved was with the mic a couple of feet away looking in the direction of where the body meets the neck. It was generally agreed that the range of sounds available from the NT1 and his guitar provided plenty of scope, and that excellent results could be achieved this way.

For his electric guitar work, Peter had had the good fortune to pick up a Line 6 Pod, in excellent condition with PSU and manual, very cheaply at a local car-boot

sale. He'd gone through the presets, but hadn't found much he liked, so I spent a few minutes setting up some '70s rock and blues sounds for him using his old faithful '70s Les Paul Standard. On our second visit, I set up four further sounds to use as generic bass DI settings for his newly acquired Aria bass, and these were based on the Pod's Brit 30 amp setting and on the Marshall-inspired amp setting played through a 4x12 speaker model. The Pod Clean amp model is also capable of some descent bass sounds, as is the Tube DI setting, though I recommended that Peter buy the update chip to convert his Pod to version two software, as that allows you to mix amp models with different speaker models.



The action of Peter's Gibson Les Paul Standard was set too high, so Paul used an old junior hacksaw blade to deepen the nut slots slightly, but suggested that Peter also have the instrument set up professionally as soon as possible.

While playing his Les Paul, I noticed that its intonation wasn't great, the reason being that the nut slots had never been cut to the appropriate depth. This meant that fretted strings went slightly sharp, and also that the action was higher than necessary. This was compounded by a slightly excessive bend in the neck, which could be redressed by a minor truss-rod adjustment. As this was potentially such a lovely instrument, I suggested Peter treat it to a proper set up at the local guitar shop, as it had evidently never been set up since it was bought, but I did deepen the nut slots for him using a ground-down junior hacksaw blade, which improved the playability immeasurably. Now when he gets it set up properly, the slots will only need a little extra dressing with the correct slotting files. We did manage to find a box spanner to fit the truss rod, but it seemed reluctant to turn and I didn't want to be the one to snap it! Maybe a good soak in WD40 would have loosened the recalcitrant nut?

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Studio Ergonomics

The recording system was arranged so that Peter had the mixer and monitors in front of him, where they should be, but his computer was directly behind him. This meant that when he was setting up mixes and effects in the computer his back was to the monitors, which is less than ideal. We felt he would be better off moving the computer to the bench to his right so that at least he could remain facing the monitors as he mixed. This would be possible if the large CRT monitor was replaced by a flat-screen model, and this would also have the advantage of avoiding interference from the monitor scan coils causing hum on his guitar pickups. On our return visit he had moved the monitor screens to the bench on his right as suggested, which made much more sense from an ergonomic point of view and also left him with more free work-surface space.



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Peter's studio layout was less than ideal for mixing in software — when seated at the computer, the main monitors were set up directly behind him.

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The Return Visit

Our second visit had to be postponed once, because Peter had managed to contract a particularly unpleasant computer virus, but when we finally arrived Peter had replaced his large VDU monitor screen with a pair of 17-inch flat-screen displays, and had also replaced his empty chocolate Hob Nob packets with full ones! We suggested that the system would work better if the Powered mixer was removed from the setup altogether, as the Allen & Heath mixer had more than enough inputs and facilities to handle recording and monitoring. As Peter currently does virtually all his mixing within *Cubase SX* anyway, the mixer doesn't actually have to do much in the way of mixing at all.

He'd rewired his system along the lines we suggested, with the direct outs on the first eight channels feeding his soundcard inputs, while the remaining channels were used to monitor the stereo soundcard outputs plus any other sources. Peter had originally been sending the mixer channels to the interface inputs via the buss faders, but, as he invariably records one channel onto one track, the direct outs offered a shorter and theoretically cleaner signal path. Peter had also bought a Powercore Element system since our first visit, but was having some problems installing it. This turned out to be due to a faulty card, and thanks to the efforts of Jim Motley at TC Electronic UK (who diagnosed the problem) and the guys at Digital Village who supplied the card, a replacement was organised without delay.

However, Peter wanted to work on some mixing while we were there, so I suggested that we use his existing Alesis Midiverb, and feed it from a spare output buss in *Cubase SX* so that it would become part of the effects send/return loop within his sequencing software. The reverb returns could be taken back to a spare stereo pair of inputs on the audio interface and then mixed back into *Cubase*, but just to prove the principle we brought the reverb return back into two spare channels on the Allen & Heath mixer, as Peter tends to mix to a stereo DAT recorder connected to the mixer anyway. This worked fine, and though not an esoteric reverb by any means, we still felt it sounded a lot better than the built-in *Cubase* reverb, which after all had to be designed to minimise CPU usage. The channel aux sends in *Cubase* could be used to control the reverb level in the usual way, and provided that Peter makes a note of the reverb settings and return levels on the analogue desk, everything should be repeatable.



border="0" alt="sos Paul@Bass +Pod3.s">

Although the Line 6 Pod is associated more with guitar sounds, Paul showed Peter how you can set up the unit to create perfectly good bass sounds as well.

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Fixing The Mixes

Since our first visit, Peter had obtained a large acoustic office divider screen which he'd placed between himself and the drum kit area to improve separation and to preserve his hearing. He'd also recorded some new material which was rock/pop in style, and the recording seemed much improved over what we'd heard before, though there were still some areas that we felt needed attention.

For example, the drum overhead mics lacked brightness, as Peter was still using his AKG C1000s for this job and they tend to have a fairly subdued high end. Adding 'air' EQ by applying a wide parametric boost at around 12kHz helped. The ideal solution would have been to assign the C1000s to duties better suited to them and to use a pair



Peter's host-based reverb processing was letting his mix down, so Hugh suggested that

of basic, flat-response, small-diaphragm capacitor mics as overheads. Where budget is a concern (and where isn't it?), there are numerous low-cost mics that will work perfectly.

he should plumb his Alesis Midiverb into the system, fed from an output of the computer's audio interface and returned to the mix via the Allen & Heath analogue mixer.

Although Peter still hadn't managed to replace the kick drum head, he had invested in a new tuning key for the rest of the kit! The continued tenure by the aged head meant the kick still needed generous amounts of 80Hz and 3-4kHz boost to add any sense of weight and impact, and even that didn't compensate for the state of the head. One tip here is that, when faced with this kind of sound at a stage where it is too late to re-record the part, you can sometimes add the desired depth by using the sub-octave plug-ins that come with many sequencers, just to add an extra layer of depth to the sound. This is really a salvage measure rather than good recording practice, but it can get you out of a tight corner when all else fails. The snare sound was also quite dull, and needed a lot of high-end EQ. Unfortunately Peter didn't have a harmonic enhancer plug-in, as I've had some success using the one in Emagic's *Logic* to add life to dull drum sounds — and indeed to lifeless overhead mics.

On one track, Peter had used a synthesised bass sound instead of a bass guitar, and to add energy and depth I tried patching it through Steinberg's tube-emulation plug-in. In isolation it didn't sound exactly like a bass guitar, but once in the mix, it played its part a lot better than previously and actually sounded pretty good. That left the vocals, which sounded absolutely fine now that Peter had sorted out his NT1 mic orientation. However, because this was a rock track the voice needed some fairly strong compression to help it sit in the mix and to give it a high-energy character. We used the *Cubase SX* dynamics for this, using a 6:1 ratio, automatic time constants, and a threshold setting that gave 10dB gain reduction on peaks. This is more than I'd normally use, but it suited this vocal part extremely well. Peter also played us some tracks featuring acoustic guitar, where he'd tried some of the mic techniques we'd explored on our first visit, and I have to say the results were pretty good and needed little in the way of EQ.

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Peter's Comments

"You would have thought that I would have seen it all and done it all, having been involved in the world of home recording for 30 years. However, when Paul and Hugh visited I was amazed at how much there was still to learn and think about — they went through the studio with a fine-toothed comb! I think I was expecting, having read previous Studio SOS articles, that I would need to cover the walls with acoustic foam, yet by the end of the session I was pleased that Paul felt the studio didn't need that treatment. I felt really embarrassed about the vocal mic being the wrong way round, but it just goes to show that, even with all that experience, you can still get something wrong! The mixing techniques I was shown were really valuable and with the new Powercore system set up I started remixing old files to bring them back to life.

"Home recording must be potentially one of the most expensive hobbies around — I imagine many readers will have felt dissatisfaction with their gear at some point, and the temptation to upgrade can seem irresistible. Replacing my monitors with flat-screen versions and buying the Powercore have burnt a hole in my pocket, and yet I've still got the drum head to replace and I must seriously think about new mics for the overhead drum positions. Oh, and the Les Paul still needs to be set up... I'm beginning to see that I need a long-term plan for buying gear, with a list of priorities. I wonder if there are any Russian football millionaires out there who would like to sponsor a small home studio? They probably couldn't afford it..."

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All's Well That Ends Well

Once again, the big picture turned out to be the sum of a lot of little parts. Rearranging the studio layout and simplifying the way the mixer was wired helped a lot, and Peter is planning to add another wiring loom so that he can mix all the separate outputs from his soundcard in the Allen & Heath mixer where necessary. Plus he has added a couple of capacitor mics to his list so that he can get a brighter overhead sound on the drum kit. His future plans include moving his control room into the adjoining garage so that he has better separation and more room for the musicians, but that's still some way off. **SOS**

Published in SOS July 2004

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SOUND ON SOUND

The World's Best Music Recording Magazine

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Studio SOS

Pete Keen and Nick Smith

Published in SOS August 2004

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People : Studio SOS

Setting up your studio in a cube-shaped room isn't a very good idea, as Pete Keen and Nick Smith found out to their cost. So the SOS team set off to Kidderminster to help find some solutions to the inevitable acoustic problems.

Paul White

Pete Keen and Nick Smith have no problems writing music or finding an outlet for it — they have just completed the soundtrack to a short film called *Just The Ticket* shown at the Cannes film festival in May this year. In fact, it was this project that convinced Pete and Nick that it was time to upgrade their studio from an early Mac G4 to a shiny new G5, because running videos within the *Pro Tools* software on their old system was, as Pete put it, like wading through treacle!

Nick works at Worcester College of Technology, which runs a well-respected music-technology course and has a couple of professionally designed studios. Nick had taken in some mixes to check them over, and he was horrified to discover that they sounded nothing like the same mixes played back in the bedroom studio. Unfortunately, this particular bedroom studio is almost a perfect cube, which



Pete and Nick had tackled the bass problems resulting from the dimensions of their studio room by building an asymmetrical false ceiling (left), containing four-inch-thick furniture foam for absorption (below). However, this treatment caused almost as many problems as it solved.

is the worst possible shape for a studio — other than perhaps a perfect sphere! Nick borrowed some acoustic panels from the college to see if anything could be done to improve the room, and after propping some of these against the ceiling using ladders and mic stands, he and Pete felt the sound was better, particularly at the bass end.

As a result of this experiment, Pete's woodworking skills were put to use building a false ceiling with holes cut into it, behind which could be wedged a generous amount of four-inch furniture foam. As Pete has connections in the furniture business, large amounts of foam seemed easy to come by, but although they made a fabulous job of the ceiling, complete with routed triangular cutouts, Nick had somehow got it into his head that studios should be asymmetrical about the monitor axis, and so he'd designed in an angle (effectively an inverted apex) about one third of the way across the ceiling. The pair had also decided to build similar structures on the two side walls, and had got as far as building the timber framing for one side when Nick called SOS for help.

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Sorting Out A Small Cube Room

Hugh and I decided this might be an interesting challenge, a decision in no way biased by the promise of chilled chocolate Hobnobs (plain *and* milk!), so we soon found ourselves in a very compact bedroom studio on the outskirts of Kidderminster. The room was approximately nine feet square, with a ceiling height of around eight feet, though the false ceiling had brought this down to around seven feet at the lowest point. Pete played some commercial records back through their Tannoy DC100 dual-concentric



The speaker drivers were still too high for optimal monitoring, even from a standing position, so Hugh turned the speakers on their sides to lower the drivers to a more suitable level.

passive monitors and explained that the ceiling had definitely helped to even out the bass end, but we all agreed that the overall sound was still somewhat muddy.

Because Pete suffers from back problems, he finds it easier to work standing up rather than sitting down, therefore the speakers had been placed on a high shelf. This resulted in the coaxial drive units being well above ear height, even when standing. Hugh cured this problem by simply turning the Tannoy speakers on their sides, therefore lowering the tweeter axis by eight inches or so. With most conventional monitors, turning them on their sides is usually a bad idea, because it corrupts the horizontal dispersion, but in the case of the Tannoy dual-concentric the dispersion is symmetrical in both the horizontal and vertical planes.

As it stood, the studio system was set up on a worktop that ran the full width of the room. It was quite congested, and it also made it difficult to make the room

acoustically symmetrical, because there was a wall full of shelves to the left of the mixing position, and a window to the right.

Although the ceiling bass trap seemed to be fairly effective in controlling the inevitably bunched standing waves of the square room, the asymmetrical design didn't do the stereo imaging any favours at all. Asking Pete and Nick to rebuild their impressive ceiling trap would clearly have gone down about as well as a plate of stale digestives, so Hugh suggested turning the studio through 90 degrees, placing the monitors either side of the window. In this way, the ceiling trap would become symmetrical as far as the monitors were concerned, and the inverted apex might even prove beneficial in terms of bouncing early reflections away from the monitoring zone. We felt that the partially built wall frames were unnecessary, and could be dismantled. Another useful side-effect of the suggested new layout was that the bench would run down the left-hand side of the room, and the shelving full of books and accessories would be at the back of the room, providing useful scattering and diffusion. The monitors could also then go on proper stands or sturdy wall brackets, and there would be room on the wall either side to affix some foam panels, taking care of side-to-side reflections at the monitoring position.



On their return, Hugh and Paul found that the whole studio layout have been changed along the lines of their suggestions, and they could then set about positioning the Tannoy monitors for the best sound — the ladder was used as a temporary stand while experimenting with different speaker positions.

The Tannoy speakers themselves sounded a little tired and flabby to our ears, and as Peter and Nick were already considering upgrading to active monitors I agreed to bring along the Mackie HR624s that I normally use as rear surround speakers to see how they worked out in that particular room. As they were mixing and processing entirely within *Pro Tools*, we also suggested that a master control box similar to a Samson C*Control or Mackie Big Knob would be useful, as it would allow them to handle headphone monitoring and the connection of two-track recorders and players without having to re-patch.

With our suggestions carefully documented, we left Peter and Nick to tackle the changes, offering to return once the supply of chocolate biscuits had been replenished...

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Apple G5 Noise Problem

Peter and Nick were very proud of their dual-processor Apple G5, but when we turned up the monitoring level I heard a very familiar noise — low-level digital 'hash' accompanied by a regular ticking. Many people have experienced this problem with connected Firewire interfaces, and it has been traced back to certain revisions of power supply fitted to dual-processor G5 models. Although the noise can be cured by using a development software tool to switch off processor idling, Apple don't recommend this, and the correct solution is to have the PSU replaced for a revision-'E' version, which the dealer or local Apple repair centre should be able to arrange under the warranty. As this particular machine was a new model provided by Jigsaw, Pete was going to contact them directly after our visit.

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Happy Returns

On our return, the studio had been transformed, with the previously cluttered bench now supporting only the necessary equipment, and an impromptu desk set up beneath the window to hold the flat-screen monitors and computer keyboard. The G5 was purring away on the floor beside the desk, and the Tannoy DC100s were standing on blocks and ladders so that we could confirm correct placement before they did anything too permanent.

The half-built wooden frame on the right-hand wall was gone, and on the side walls they'd fixed some more furniture foam, topped by some Auralex panels — the extra thickness of foam would absorb to a lower frequency, though it probably wasn't essential. They'd also finished wiring the ceiling lights and crammed even more foam into the ceiling to improve its effectiveness at low frequencies. Finally, because the studio had been rotated by 90 degrees, the newly built ceiling was now nicely symmetrical about the monitoring axis.

The sonic improvement surprised even us, as cube-shaped small rooms are definitely bad news for audio accuracy, especially when they are this small. What we discovered was that the bass was very even and consistent everywhere in the room apart from a beach-ball-sized volume, which we nicknamed the 'zone of death', exactly in the centre of the room — here, the bass just disappeared completely. However, provided that this zone was avoided, the monitoring seemed pretty accurate and reliable everywhere else (except very close to the walls, of course). The stereo imaging was astonishingly good too.



Moving the Mackie speakers to the window sill demonstrated how much the placement of the speakers affected the sound — the bass end immediately became much less reliable.

Next we tried out the Mackie monitors. The bass end felt even more controlled and the high end detail was more revealing than with the Tannoy passive speakers, and both Nick and Pete agreed that they were now hearing things in their mixes that simply weren't audible before. Although Peter wanted to use wall brackets to hold the speakers, I strongly recommended stands, as they give you the opportunity to fine-tune the speaker position for the best results. After seeing how easy it was to build our SOS DIY stands (as shown back in April's Studio SOS article), he resolved to build his own from timber.

Just to demonstrate how important speaker positioning is, we moved the Mackie monitors from either side of the window onto the window ledge. The sound changed dramatically — for the worse at the bass end. A little further experimentation confirmed that stands either side of the window, but away from the room's corners, would be difficult to improve upon.

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What Nick & Pete Had To Say

"Initially we had attempted to treat the ceiling with 12mm MDF backed with 100mm high-density CM40 sofa cushion foam. We cut triangular holes in the MDF using a jigsaw, providing a 3.5:1 ratio of solid to hole, and cleaned them up with a router. The panels were then supported on bearers on opposite walls and angled down to meet on a beam suspended 300mm from the ceiling two thirds of the way across the room. This helped to even out the bass end, but the overall tonality and general stereo imaging still left a lot to be desired, which is why we asked SOS for help.

"Because our new Apple G5 and Digidesign Digi 002 system was still in its box, Paul and Hugh suggested we re-organise the studio based on their recommendations and set up the new equipment ready for a return visit a few weeks later. Once we'd overcome a few teething technical difficulties, everything seemed fine, until Paul brought in his Mackie HR624 active monitors, which revealed a strange low-level digital noise that Paul instantly recognised as being attributable to the G5 power supply, so that now needs to be sorted out.

"The Mackies turned out to be perfect for our room, adding extra definition and clarity to the bottom end, so they're at the top of our shopping list. No amount of experimentation could rid us of the 'zone of death' in the centre of the room, where all the bass disappeared, but at least we know to work around it. As long as we stay out of the centre the difference in the evenness of the bass is amazing, and the stereo imaging is now excellent. Even with our current Tannoy DC100 monitors we can now mix with confidence.

"Improving the acoustics may take a little time and money, but in this case it has been a far better investment than a computer upgrade or new microphone. Had this article been published before choosing our studio we would definitely have reconsidered our choice of room, and done some experimenting with the acoustics before jumping in with both feet. Although it eventually cost us two whole packets of chocolate Hobnobs, it was worth every crumb!"

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Box Clever

This Studio SOS visit showed that, although small square rooms are generally bad news for audio, they can be made usable by applying appropriate acoustic treatment, provided that you're careful about where to sit and where not to sit when mixing. Pete and Nick are planning to use this studio for a couple of years while they have a garage built, which will free up enough space for them to do the job properly. While I'd be the last to claim that they now have perfect acoustics, the improvement is enormous — they can at least hear the detail and stereo imaging of their mixes and make a fair stab at judging the level of bass in a mix. They shouldn't try to master their own mixes in this room, but that doesn't mean that they should have any problems doing mixes for fine-tuning at the mastering stage. **SOS**

Published in SOS August 2004

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SOUND ON SOUND

The World's Best Music Recording Magazine

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Studio SOS

TV Composer David Lowe

Published in SOS September 2004

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People : Studio SOS

This month, we help television composer Dave Lowe transform a cavernous-sounding spare bedroom into a usable home studio.

Paul White

David Lowe is one of today's more successful writers of TV music. He is probably best known for the current BBC news theme, though he has also produced some excellent hard-hitting drama themes, such as *Mersey Beat* and, of course, the pop single 'Would You?' under the name Touch & Go. You may also recall that some months back he agreed to let me co-write the *Royal Gardeners* theme with him, so when he asked for help with his studio I could hardly turn him down!

He needed help to turn a bedroom in his Spanish villa into a studio that could be used not only for writing but also for tracking and occasionally mixing. Although his main studio is near Malvern, David already has a duplicate setup in his London apartment and he wanted to do the same in Spain, so that he could continue working on his TV and album projects when away from home. Like most villas in the area, David's has hard tile floors, solid rendered walls, and (in the case of his studio room) flat terra cotta tiles on the sloping ceiling, below the main roof tiles. This gave it a sound rather like an '80s live drum room: very reverberant, and far too resonant to mix in.



The upstairs room he'd chosen had patio doors looking out onto a veranda, but also had a two-metre opening looking out onto an adjacent room, which he felt might be useful for recording vocals. The studio room itself turned out to be fairly small, measuring around 3 x 2.5m, but the shape was also somewhat awkward, because of the large opening into the next room. To complicate things further, the adjoining room contained a door leading to a stairwell, and if this was left open the stairwell reverberation made its presence known in no uncertain terms!

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Location, Location, Location...

Finding space for the equipment was not a problem, as David wanted to keep this studio simple. The equipment comprised only a dual-processor Mac G4 running Emagic *Logic Pro*, a Logic Control with two expanders, and a MOTU 2408 interface. An Emagic AMT8 was in place to handle the MIDI (though we had to make do with an MT4 and just the main Logic Control unit, as David had inadvertently left the AMT8's PSU at home), and an ageing Korg M1 was set up as a master keyboard. This didn't have the audio outs connected, because David intended to use only software instruments. As it turned out, we were just able to slide the M1 on its 'X'-frame stand under the rear of the desk so that only the keys protruded — a useful tip if your 'X'-frame stand won't go low enough for this is to dispense with the securing crossbar and instead use a strap or length of strong nylon cord to hold the top two supports the correct distance apart and therefore provide the necessary height.

A Mackie 1202 mixer was used as a front end for recording, and the sole studio mic was a Neumann U87, mounted in a shockmount with a pop shield. Monitoring was courtesy of a pair of Dynaudio BM15s fed from a Samson 550 Servo power amplifier, where the monitors were set up on metre-tall stands and connected with adequately heavy speaker cable.

Not knowing what supplies were available in the area, I studied photographs of the studio in advance of the trip and then suggested some materials for David to order and have shipped over. Happily these all arrived before I did! I planned to use simple Auralex foam to treat the walls and corners, augmented by a pair of Real Traps Mini Traps suspended from the ceiling to



The first step towards reducing the excess reverberation in Dave's room was to install several Auralex acoustic foam panels and corner traps using contact adhesive spray.



A large area of foam treatment was created by sticking multiple foam panels to MDF, and this

deal with reflections from that area and also to provide additional low-frequency trapping. I felt that, as far as bass trapping was concerned, the large window area in the room and the one adjoining it would help us, as would the wooden cupboards in the next room. Bass tends to pass through windows rather than getting reflected, so the more window area you have, the more low-frequency energy is lost to the outside world. Cupboards with panelled wooden doors also help damp down bass modes, especially if they contain towels and bedding.

could then be hung on the wall like a picture.

When I arrived, David had already set up his gear on a table in what he felt was the most appropriate arrangement. This had him looking into an alcove, with the dividing wall between the two rooms on his left. The patio door to the outside was on his right, and the wide opening to the next room was behind him and to his left, so there was no semblance of acoustic symmetry. Before looking at the acoustic problems, I suggested to David that he face his system the opposite way, so that he would be sitting in the only part of the room where he could have a wall on either side of him. This would provide a degree of symmetry, if only in the listening position, and it would avoid obstructing the patio doors. As a bonus, it would also afford David a better view of the outside world during those valued moments of reflection! We moved the table to see how this would look and David agreed it felt better, so we decided to continue work on this basis.

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Acoustic Treatment

The Auralex corner traps went in the two corners behind the mixing position, though these couldn't be exactly symmetrical, given that the roof was higher at one side of the studio than at the other. Also, the lower of the two corners housed some electrical fittings and switches that needed to be left exposed. The traps were glued directly to the painted wall using the included spray contact adhesive — you have to position the traps very carefully when using that stuff, though, as once the glue grabs there's little chance of it letting go again!

To avoid doing anything too permanent to the rest of the walls, the remaining four-inch-thick Auralex panels were glued to MDF sheets and then hung on the wall like pictures, mostly using existing screw holes and wall plugs that had been fitted by the previous occupant to fix hooks. Strong nylon cord was passed between holes drilled into the MDF before the foam was glued in place. Our trip to the local equivalent of B&Q turned out to be interesting, as we not only had to buy all



After a trip to the Spanish equivalent of B&Q to purchase the necessary fixings and tools, the Real Traps Mini Traps panel absorbers were rigged to the ceiling of the studio room using hooks and fine chain.

the necessary materials, but also basic tools, such as a wood saw and a hacksaw — however, this was achieved fairly painlessly with the aid of much pointing and drawing of diagrams!

Because the room comprised entirely reflective surfaces, I decided on a vertical panel at either side of the listening position with a further sheet suspended horizontally behind the chair at head height. One of the vertical panels was suspended from a curtain rail, as there was a narrow window in the wall to the left of our new mixing position. As the wall behind the monitors was flat and completely bare, I used the remaining three panels of foam to treat that, again fixed to MDF sheet and hung on hooks.

The Real Traps Mini Traps work on a different principle, as their performance improves if they are positioned away from a surface rather than directly on it, but in this room it wasn't practical to place them in their optimum position for bass trapping, which is across a corner or wall/ceiling junction at around 45 degrees. Instead I suspended them from chains and hung them from hooks screwed into the ceiling joists, so that they would hang parallel to the floor



above the listening position and the desk. This way they would intercept any ceiling reflections that tried to reach the listening position and would also provide additional low-frequency trapping, albeit not as effectively as if they'd been hung across corners. I was surprised at how hard it was to cut what was in effect plug chain, and in the end I had to use a hacksaw, as none of our pliers or wire cutters would touch it!

We'd also ordered a pair of Auralex Max Walls, which are foam panels mounted on included stands to provide movable treatment. We placed these in front of the large window in the adjacent room, as David hadn't yet had the blinds made for this room, but the real reason for buying them was to create a localised area that could be used for recording vocals and instruments. David had felt that the adjoining room would make a suitable playing area, but because of the open-plan nature of the two rooms and the amount of noise kicked out by the G4, I suggested that the next bedroom along would be better for this purpose, as it had a double bed in it (which made it sound less lively) and there was space under the door for a mic cable.

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Monitoring Adjustments

Once the acoustic treatment was up and the equipment put in place, we found ourselves short of UK mains sockets. David had one spare four-way plugboard with a very short cable terminating in a European plug, so I attacked a UK IEC

lead and used it to replace the existing short cable. A European plug adaptor was then used so that we could plug it in. Shortly after this, David said he was getting a tingle when he touched his Logic Control, which revealed an interesting fact about European mains connectors. Although these plugs look like two-pin devices (on account of them having two pins!), they also have a recessed earth connector on one edge that contacts a matching connector in the wall socket. However, you can insert these plugs either way up, as the socket has grounding contacts on both sides. In this case, one of the contacts had lost its springiness and so wasn't connecting to the plug. By inverting the plug, the earth was restored and the tingling stopped, but replacing the faulty wall socket would be the only safe long-term option.



There was quite a lot of noise coming from Dave's Apple G4 computer, so it was moved further away from the mixing position. Although this helped, a further improvement was achieved by draping a blanket over the top and sides of the computer, leaving the ventilation unhindered, with a piece of acoustic foam placed fairly close behind the machine to soak up a bit more sound.

Before doing any listening tests, we had to position the monitors correctly, and with the stands David was using the speakers were around 30cm too high. The answer was to use some foam offcuts as wedges to prop up the rear of the each speaker so as to angle it down and aim the tweeter correctly. Non-slip matting, from our friendly Spanish hardware shop, was used to stop the speakers sliding off their stands. The BM15s have different left-hand and right-hand speakers, so I set them up with their tweeters outermost to give the widest stereo image. The foam plugs fitted to the bass ports (these come with the speakers) were left in place, as it turned out that there was absolutely no shortage of low end.

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David's Comments

"I must admit to being a bit sceptical about what could be achieved in that room without spending loads of time and money filling in ceilings, knocking down walls, putting in new doors, and so forth. But it had all the makings of a fantastic room to work in, if we could get rid of the huge cave-like reverberations! Paul has worked wonders, achieving far more than I imagined could be done, and all with only a few simple but effective materials, all easily available at comparatively modest cost. When I sat in and listened to some of my tracks in the finished room, it was almost like having headphones on! An added bonus was Paul's idea to move the setup around the other way, something I'd never have thought of myself, but which was a tremendous improvement. I also loved the tip about lowering the keyboard stand — definitely worth the price of a Spanish packet of Jaffa Cakes!"

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Success?

Checking the audio playback confirmed that the original ringy, splashy space had indeed been tamed to a surprising degree. The bass end was adequately even and there was now much better stereo imaging. The flutter echoes were gone, and talking in the room felt more like talking in a studio rather than in a deserted warehouse. I wouldn't go so far as to say we now had a room you could mix and master in with complete confidence, but at least it was fine for tracking and perfectly adequate for mixing provided that you'd have a chance to master the material elsewhere. David agreed that the improvement was vast, and he also felt that the room had a better working ambience, with a more intimate vibe to it.

A further acoustic improvement was realised when David hung some heavy made-to-measure curtains that his wife Helen had sent over for him. These went over the patio doors and the opening between the two rooms, but they improved the sound even when open. We tried to source some rugs for the floor, but rug shops (and rugs) seem very thin on the ground in Spain — although the room sounded perfectly usable without rugs, adding them to the studio and adjoining room would doubtless help dry up the sound a little more. I also suggested putting slatted wooden blinds over the very wide window in the adjoining room in order to scatter sound and let in light at the same time.




Paul White, Paul White...He's a secret Jaffa Cake eater...

Of course all this new-found peace and tranquillity was disturbed by the racket of the G4 whirring away on the floor, so I decided to try moving it into the adjacent room where the dividing wall and new curtains would offer a degree of screening. A USB hub was pressed into service to extend the keyboard/mouse connections, and the monitor cable was just long enough.

A simple sound-deadening trick I discovered when I had a G4 was to place a folded blanket or rug over the computer to form a tunnel, leaving the front and rear of the machine exposed to the air. The blanket needs to go all the way to the floor to be effective, but it is essential not to block the ventilation path to the front and rear of the machine. We tried this, adding the spare Auralex bass trap behind the G4 to try to soak up a little more sound. The drop in noise was significant, mainly due to the folded rug and the new computer position. Although still not completely abolished, the computer noise was now barely noticeable from the engineering position.

Our last day was spent doing a spot of recording to check out the system, which happily confirmed everything was working fine and that there were no ground-loop hums or other nasties. Even without rugs and blinds, the studio sounded perfectly workable, though it was important to keep the door to the stairwell closed. The only down side was not being able to have Hugh along on the trip,

but out of a sense of tradition and loyalty I made sure to eat his share of the Jaffa Cakes! 

Thanks to Audio Agency and Sonic Distribution for arranging prompt shipment of the acoustic treatment products used in this article.

Published in SOS September 2004

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SOUND ON SOUND

The World's Best Music Recording Magazine

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Studio SOS

Dorian Kelly

Published in SOS October 2004

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The SOS team ride to the rescue of a budding media composer who's having trouble with his mixes.

Paul White & Hugh Robjohns

Dorian Kelly has long-term plans to compose music for TV and film after successfully completing a media composing course (the home-study Music For The Media course run by Guy Michelmores from Atlantic Studios), but in order to gain experience he has set up his own Steinberg *Cubase* SX-based home studio where he works on every type of music possible, from classical to pop. This seems to be a very sensible and practical approach, but although Dorian's room has had the benefit of some acoustic treatment, he was worried that he couldn't hear enough low end in the studio, which left him with mixes that sounded bass heavy when played elsewhere. He was also keen to try to improve his mixing technique, so he called SOS.



Dorian's monitors were set up on shelving, which was not rigid enough for the purpose and which therefore compromised the bass response of his Spirit Absolute II monitors, speakers which already have limited low-end extension.

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Low-end Monitoring Problems

When we arrived at Dorian's Reading home, where his studio is located in a converted garage, the kettle was boiling and Dorian had assembled a vast plate of chocolate digestive biscuits and croissants — so he was clearly aware of all the necessary Studio SOS protocols! While attempting to reduce the size of the cookie mountain, we played a CD of various test tracks over Dorian's system,

which comprised a small Behringer mixer to handle monitoring sources, a Samson power amplifier, and a pair of passive Spirit Absolute II speakers. The recording system was based on a purpose-built PC running *Cubase SX* with another small Behringer mixer to manage the input signals. An Emu Proteus 2000 provided the only external MIDI sounds.

As we have discovered on previous occasions, the Absolute II monitors don't have a great deal of low-end extension. This, combined with the room size and shape plus 'less than ideal' speaker mounting did indeed produce a bass-light result, though the mid-range and high end were fine. While Dorian's room is adequately long, it is only a couple of metres wide, and in a room this shape there's little alternative than to work across the width of the room, as there would be insufficient space to set up the equipment in an ergonomic fashion if working across the narrow axis of the room.

The wall behind the mixing position was already treated with some acoustic foam tiles and a few small areas of thick carpet tile, while the ceiling was largely covered in fibrous office-style acoustic tiles — the overall result was reasonably well controlled, without any obvious boomy bass notes. The plasterboard walls, a cupboard, the doors, and two windows all contributed to some fortuitous low-frequency trapping, so the main problem was the reduced level of bass from the monitors, rather than uneven bass.



Even when Paul switched the Absolute IIs for his own Mackie HR624s, which have a better bass response, some adjustment of the speakers' onboard frequency tailoring switches was required to give a representative low-end picture.

Dorian had his speakers set up on foam speaker pads that rested on the free-standing pine shelving he used to store all his studio accessories, both above and to either side of the mixing position. Using these foam pads is preferable to standing the speakers directly on non-rigid shelves, but there's no real substitute for solid stands or wall brackets. However, in this case these couldn't be accommodated for space reasons. To evaluate the room further, we set up our Mackie HR624 active monitors in place of the Absolute IIs and played the test CD again. The result was better, but we still weren't hearing as much low end as we are used to from these speakers, so the room was clearly a contributing factor. Setting the bass switches on the monitors from their half-space mode to full-space mode delivered a better balance, even though the speakers were being used close to a wall where the half space setting was technically correct. The full-space setting is normally only used when the speakers are set up away from walls, but as it worked we weren't going to knock it! After all, getting any kind of decent monitoring system working in a room this narrow is inevitably a compromise.

After hearing the system with the Mackies in place and after playing some of his

own mixes through them, Dorian agreed that he should change his monitors as soon as possible, but because of budget constraints he felt he had to look for something rather cheaper than the Mackies. He wanted to go for active models, and after looking at the options he narrowed his choices down to Alesis, KRK, and Fostex units.

On the subject of mounting the monitors, the existing position was just a little too high, as the tweeters should ideally be aimed at the engineer's head and not a few inches above it. This could be remedied by either lowering the shelves or by angling the speaker downwards and inwards, but it would also be beneficial to provide a more stable mounting platform. You could use a metal bracket to secure the speaker shelf to the wall behind it and then position a small paving slab (around 300mm square) as a base for the speaker — the additional mass of the slab helps to damp and control vibrations. You might also want to stand the speaker on the slab using blobs of Blu-Tac under each corner rather than relying on foam. In a fair and just world, the 'slab' approach should tighten up the bass end to a worthwhile extent.

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Vocal Recording & MIDI Tips

Dorian played us some vocals he'd recorded where the singer had stood at the opposite end of the room from the computer and sung with the back of the mic facing the mixing position to minimise noise pickup from the computer. The results were reasonable, but some room coloration was evident and, as vocals often need compressing to make them sit properly in a mix, the room ambience is usually emphasised further. Dorian had bought one of those Auralex foam gizmos that clip over the back of the mic to attenuate sound coming from that direction, but the wall directly behind the singer was fairly bare and had a mirror mounted on it, all of which conspired to bounce sound back into the microphone.

A practical solution to this is to hang a double duvet from a rail fixed to the ceiling, so that it is directly behind the singer when recording. In this way, ambient sounds no longer reflect back into the front of the microphone, and the room coloration is greatly reduced. If the duvet can be hung from a curved rail so that it also comes around the sides of the singer slightly, then so much the better. Dorian liked this idea, as it was simple and inexpensive and the duvet could be pushed to one side when not needed provided that it was hung using curtain rings.



A combination of two reverbs in series was required to get the best out of *Cubase SX*'s built-in plug-ins: the first (left) added a short brighter ambience to give a sense of roominess; and the second (right) contributed a subtle reverb tail without clogging up the mix.

Next Dorian played us some of his classical-style compositions, which had been

created using Steinberg *Halion* samples combined with some tracks of him playing real trumpet and French horn. Overall, the sound was well balanced, though we did identify three specific problem areas. The most obvious problem was that some of the parts sounded quite stilted, and this turned out to be because Dorian composed his music using *Sibelius* before transferring it into *Cubase SX* to mix. As *Sibelius* creates MIDI files based on the bars and notes of the score, the data is inherently tightly quantised, but in addition Dorian wasn't making use of those instruments that supported legato mode to allow overlapping or consecutive notes to flow smoothly into each other. Selecting the desired MIDI notes in the *Cubase SX* grid edit page and then choosing Legato from the MIDI menu removed spaces from between the notes and allowed the sound to flow more naturally, without constant re-triggering of samples. This is particularly evident on string sounds (which 'suck' unnaturally if legato isn't used correctly), but solo wind instruments and bowed solo strings also tend to play legato a lot of the time, so trying to simulate a performance from distinctly separate notes will invariably sound mechanical. We tweaked a couple of parts to demonstrate the difference when parts were played legato, and Dorian immediately saw the advantage.

Some instrument patches are created specifically with legato playing in mind, and some instruments have dedicated legato trigger modes which prevent the envelope from re-triggering when a new note is played, provided that it overlaps the previous one. These techniques are all worth exploring, as is the trick of using a MIDI volume pedal (transmitting MIDI Continuous Controller number seven) when recording strings to create more natural-sounding swells and crescendos.

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Reverb Overhaul

The other obvious shortcoming in Dorian's tracks was the rather limited reverb that comes with *Cubase SX*. Although the mix was obviously reverberant, there was no sense of space or acoustic — an obvious failing in the context of orchestral music. To improve on this, Dorian has the choice of buying a convolution-based reverb such as *Altverb* (which will eat up a lot of processing power), adding a hardware DSP card such as the TC Electronic Powercore or Universal Audio UAD1, or plumbing in an external hardware reverb processor such as a TC Electronic M*One XL or a Lexicon MPX550. Because Dorian's M Audio Firewire Audiophile interface has two analogue inputs and outputs, plus S/PDIF in and out, a hardware reverb with digital I/O could be plumbed into the system via S/PDIF to save unnecessary conversion. Any latency affecting the external reverb would be negligible compared to the usual pre-delay added to reverb — provided that the reverb was used in an aux send/return loop and set to 100 percent wet.

However, my own preference would be for a card-based system, as you get the quality of hardware but the effect settings are stored as part of the *Cubase SX* song. So when you revisit old projects, you don't spend half the day trying to get

back to where you started. You can also run multiple instances of the reverb plug-in with a card-based system.

As an experiment, however, we decided to see what could be achieved using the *Cubase SX* native *Reverb A*, and the strategy we tried was to use two reverbs in series, the first set to as short a decay time as possible to simulate an early-reflections/ambience type of treatment. This was adjusted by ear to give a coloured roominess to the sound (we ended up with almost as much reverb as dry signal) and a short pre-delay of between 10-20ms seemed to help. The ambience reverb was followed by a longer concert hall effect with a 70-80ms pre-delay to provide the tail of the reverb. We set this at a much lower level so that it added a subtle reverb tail without choking the gaps in the music, and the result was surprisingly good.

Obviously it was not as smooth as a real high-end reverb, but at least we managed to create a sense of space and depth without clogging up the mix with an excessively loud reverb tail. The use of an ambience-like reverb to create the basic sound character certainly helped the overall sound become more homogeneous, and the performance now sounded as though it had taken place in a real space.

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Processing The Final Mix

The final challenge was to process the final mix to create a more commercial, 'TV-friendly' sound. Dorian had tried some fairly radical EQ that combined shelving bass boost with a wide mid-range cut to generate something like the traditional smile curve (more of a manic leer really!). However, Hugh and I felt that using only the two parametric sections of the *Cubase* four-band EQ would give us more control, so we added a gentle 80Hz hump balanced by a broad 13kHz boost to add air and sizzle to the top end without making the mid-range sound harsh.

This was teamed with a compressor setting using a low 1:1.2 ratio, a fairly fast attack, and the automatic release setting. The threshold was turned down until the gain-reduction meter showed around 5-6dB on the louder passages, which left us with a threshold setting of around -35dBFS. This strategy reduces the whole dynamic range of the material in a very gentle way, and differs from the more usual tracking approach where a higher threshold is set and then everything above the threshold is squashed much more aggressively.



With the Mackie monitors in place, Paul had a look at the processing Dorian had used on his final mixes. His EQ settings in particular were felt to be a little extreme, so a gentler 'smile curve' EQ was used instead.

Dorian then played us a pop song he'd recorded with a female vocalist. Her performance lacked a little conviction, but was basically sound, and we tried to treat it with a more robust 8:1 compression ratio, the fastest attack, and the automatic release mode. Again, the threshold was adjusted to show the desired amount of gain reduction, in this case around 8dB on peaks. This worked fine, making the vocal sit better in the mix and also making the sound appear more confident. Dorian was a little unsure about the best reverb treatment to use so we again tried the approach of layering a short pre-delayed bright setting to create an almost slapback ambience together with a longer reverb tail at a lower level and with more pre-delay. After a little tweaking we arrived at something more contemporary sounding, where the vocal took command of the mix, rather than apologising for being there as it had before! A little high EQ to sharpen it up and we were done.

When discussing how the vocal was recorded, Dorian also admitted that he hadn't added any reverb to the monitor mix, though Cubase SX does allow this and it can help make a singer feel more confident. However, you do need to check with the singer how much reverb they like to hear in their cans, as everyone seems to have their own preferences in this area. Too much is just as bad as too little.

It can also make a difference to reverse the polarity of the headphone feed while recording, which can be achieved by using the phase-reverse button in the dynamics section of the vocal channel in the case of *Cubase SX*. This affects the way the singer hears the sound from the headphones when it combines with what they hear of themselves directly, and usually one polarity sounds more comfortable than the other.

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Dorian's Comments

"Despite feeling a little nervous of what Paul and Hugh would make of my setup and my music, it was great to have the SOS team here. Being a musician first and an engineer second, I do not always find it easy to resolve technical problems I encounter. Paul's Mackie monitors made a big difference, and after hearing them in action I will definitely invest in a pair of active monitors in the near future (probably the Alesis M1 Active MkIIs). It's very reassuring to know that there is not too much wrong with the room itself.

"Paul's advice on the MIDI files will be very useful. Most of the time I am trying to recreate a live music sound by combining some audio tracks (such as trumpet, French horn, and violin) with samples triggered over MIDI, and this is often more difficult when going for an orchestral



sound. Using Paul's tips to make these sound more realistic will be very useful. If I had more space and some kind of budget I would use live musicians all the time, but at the moment that just isn't practical.

"Out of all the great information that Paul and Hugh passed on, I think the most useful for me will be the compression, EQ, and reverb settings. They managed to improve my mixes enormously in a very short space of time, just using the bundled plug-ins within *Cubase SX*. I was particularly impressed with the reverb settings for the orchestral mix — layering two different reverbs was a great idea, and one I would not have come up with myself. Combined with subtler compression and EQ, it made the mix sound much more natural — almost like it had been processed 'invisibly', which is very important when creating convincing orchestral tracks. I am currently in the process of creating show reels to send out to every man and his dog in an attempt to actually get some work, and after the SOS visit I feel a lot more confident — I'm sure my CDs will now sound more professional as a result."

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Optimising The PC

Towards the end of our visit, Dorian asked a few questions from a checklist he'd been compiling before our visit. The first was about latency and buffer sizes. He wasn't quite sure what to adjust or what the best setting was, but he was aware that it was important. Latency is usually defined as the time delay between a signal being piped into a computer workstation and that same signal appearing at the monitor outputs. It is most important in the context of software instruments, as the monitoring workarounds that can be tried when recording (monitoring the vocal mic at the input to the system rather than at the output, for example) can't be used with software instruments. If the latency is too long, the time lag between pressing a key and hearing a sound will put you off playing, but on the other hand if you try to set up your system for a very low latency, the essential data buffers in RAM may be too small to allow a continuous flow of audio data, and that's when you get clicks and pops.

On a modern computer like Dorian's (which has 3GB of memory), it is usually OK to set the buffer sizes to 128 or 256 samples, which at a 44.1kHz sample rate will give a low enough latency that most users will be unaware of it, yet the buffers are still large enough for reliable operation. Of course the stress on the system gets worse as you add more plug-in effects and instruments, so when the computer is pushed close to its limit, it may be necessary to increase the buffer size to avoid glitching.

To minimise the CPU load, a handy tip is to scale the song display so you can see it all on one screen, which avoids the computer having to work on screen redraws during playback and when mixing. You can also increase the buffer size when you're mixing, as latency isn't an issue unless you're recording. The other useful strategy is to render instrument tracks as audio files, which saves on CPU load at the expense of a little more hard-drive activity, but in my experience most modern systems are capable of handling far more audio tracks than any sane composer would ever need unless they're scoring a Hollywood movie and creating all their ensembles by layering individual monophonic tracks. There are

many articles discussing the causes of latency and its workarounds on the SOS web site, so it isn't necessary to go into more detail here.

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All Of A Dither

Dorian's final question involved dither. Again, he had heard that it was a good thing and was beneficial when reducing the resolution from 24 to 16 bits, but he wasn't sure where or when to apply it. In this context the reality is that dither is only of benefit if it is the very last thing you do to a piece of audio, so it's no good dithering your mixes into 16-bit files, putting them into a CD playlist, and then changing their relative levels — the act of adjusting the levels will destroy the dithering benefits. My own strategy is to record and mix everything at 24-bit resolution, then use a CD playlist compiling program (Roxio Jam for Mac in my case) that automatically applies dither during word-length reduction as the very last stage after any other adjustments you may have made before burning the CD-R.

At the end of the afternoon, we felt we'd made a worthwhile amount of progress. We'd demonstrated that the room acoustics were reasonably good but that the current choice of monitoring wasn't ideal for the room. We'd also come up with some cheap and cheerful ways of improving the recorded vocal sound, and our impromptu dual-reverb treatments had worked out better than expected. We had also covered some useful compression and EQ basics, as well as made some suggestions for tweaking MIDI files to make the playing sound more realistic — specifically by using legato. The job was done, the plate of biscuits was looking decidedly well hammered, so we packed up our toys and made for home. **SOS**

Published in SOS October 2004

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SOUND ON SOUND

The World's Best Music Recording Magazine

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Studio SOS

Tony Global

Published in SOS November 2004

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People : Studio SOS

The SOS team help a reader in Nottingham to polish his vocal sounds and improve his mixes.

Paul White & Hugh Robjohns

Tony Global's background is in the dance-music/DJ scene, releasing records under various guises for the Tummy Touch and Breakdown labels, but since doing production work for singer-songwriter Matthew Jay a few years ago (a collection of his early material entitled *Too Soon* has just been made available on www.matthewjay.com) he has increasingly been working on guitar-based music. His current project is producing an album of what might loosely be described as electric/acoustic melodic folk/pop, which features some excellent vocals from talented vocalist and songwriter B Kristiansen — they intend to release this on their newly-formed label Urban Folk. His studio is in a converted loft space above his Nottingham home, where he has a Mac G4 running *Logic*, PMC TB1 and Yamaha NS10 monitors, and a wonderful private collection of classic recording gear. He also seems to have most of the old hardware synths of which Paul now has the plug-in versions! A MOTU 828 is used as the audio interface to the Mac, while a Mackie 1604VLZ Pro mixer is used as a routing and monitoring station. Acoustic guitar parts are recorded on two tracks using an AKG C414 and the guitar's built-in pickup via a



Tony's PMC monitors were moved wider apart to improve their stereo imaging, and the driver mounting bolts of his NS10s were tightened to reduce the potential for rattling and detrimental air leaks.

Drawmer 1960 preamp and compressor. Most electric parts were input via an early '60s Binson Echorec valve delay unit (used as a preamp for warmth and tone) and the Drawmer 1960.

Like many people with computer-based studios, Tony has found that he's relying more and more on software and less on hardware, and since upgrading to *Logic Pro Space Designer* is being used in place of a hardware reverb. Tony called us in because he felt that he could improve both his recording technique and his mixes, and he also wanted to check out his monitoring environment. The bait was a selection of chocolate comestibles and real coffee!

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Stereo Image & Phase Problems

Although his studio is small and loft-shaped, with an apex roof, the combination of a soft sofa at the rear of the room and lots of stored equipment seemed to tame the acoustics pretty well, and the bass end proved to be reasonably even when we played our test CD over the PMC nearfield monitors. However, the stereo imaging from the PMCs was not up to the high standards that Hugh knew them to be capable of, probably because they were placed relatively close together on an intriguing pair of concrete speaker stands. So after loosening the cables from the cable clips behind the desk, we swapped the PMCs and NS10s over, putting the PMCs on the outside of the monitor pairs. This delivered the expected improvement in imaging, and we felt the system provided a good, reliable monitoring environment.

The PMCs were powered by a Bryston 3B amplifier, via a switch box that allowed the speaker signal to be routed to either the TB1s or a pair of NS10s. So having checked out the PMCs, we switched to the Yamaha monitors and realised instantly that they were wired out of phase. Tony had just undertaken some major rewiring and hadn't really used the NS10s, as the PMCs are his main monitors, which was why he hadn't yet noticed. The signs were unmistakable though — it was like having your ears slowly sucked out by a large sink plunger! After tracing the wiring it became clear that the error was in one of the NS10 connections at the back of the speaker switching box, so Hugh clambered underneath the bench with a screwdriver and remedied the fault.



Tony and B helped Paul temporarily place some acoustic foam on the ceiling to see whether ceiling reflections were compromising the monitoring sound.

Tony had also reported distortion from the NS10 speakers when they were turned up loud, but this couldn't be reproduced after Hugh had fixed the wiring so it may have been down to a poor contact, or even the way Tony perceived the

effect of the phase reversal. However, another common source of rattles, air leaks, or other distortion-like noises can be if the bolts that hold the drivers into the loudspeaker cabinets work loose, as they tend to with age and use. To be sure that this wasn't a problem, Hugh and Tony set to with screwdrivers and Allen keys to make sure everything was tight and secure.

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Fine-tuning The Acoustics

Tony initially phoned us for advice on setting up his studio a few months ago and had implemented many of our suggestions before we arrived, so at least everything was in the right place and the monitors were standing on thick concrete slabs atop sturdy concrete stands. The only change we made was to position his PMC speakers on the outside and the NS10s on the inside to produce the best stereo imaging from the best pair of monitors. Both sets of monitors were fixed with blobs of Blu-Tac to the concrete slabs and were angled inwards towards the listening position. We re-checked the sound by playing a selection of test tracks we take around with us for exactly this purpose and, considering how small the room was, we were pretty happy with the results.

In fact the only improvements we felt we could make were to put some acoustic foam behind the monitors, a further patch on the ceiling near the apex to kill ceiling reflections, and additional foam at either side of the monitoring position to kill side-to-side reflections at mid-range and high frequencies. This is desirable as it helps to tighten up the stereo imaging, and as we had some spare Auralex panels with us we propped these in place temporarily to prove the point, then left them with Tony to fix more permanently later.



The Yamaha NS10 monitors were out of phase, so Hugh set about finding the cause of the problem. In the end the fault was traced to the monitor switcher mounted under the desk — the NS10s had been incorrectly connected.

Because the Auralex panels we had available clashed pretty violently with the room's colour scheme, we suggested that Tony build them into a simple wooden frame which he could then fix to the wall, and cover the whole thing with a light-weave fabric of an acceptable colour. The panels would be just as effective acoustically, but aesthetically more pleasing and much easier to remove (when the time came to move house) than if the foam was glued directly to the walls in the conventional way.

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Vocal Recording & Processing

We then turned our attention to some tracks that Tony had been working on, and it has to be said at the outset that the general standard of recording and mixing was very good, and there were some great performances and musical arrangements. However, Tony was not quite happy with the way B's vocals sat in the mix, and as he'd found most reverb treatments to be too wet-sounding, he'd opted on the side of keeping them pretty dry, which had the result of making them sound slightly displaced from the rest of the mix.

He had been recording B with her facing a corner of the studio which had been festooned with blankets to make it sound less lively. However, this arrangement has the mic facing out into the room, and so it captures the room ambience. We suggested making future recordings with her standing with her back to the corner and singing outwards. That way, the cardioid pattern on the AKG C414B ULS mic Tony was using would inherently reject most of the unwanted room ambience, and anything bouncing off the wall behind the singer would be soaked up by the blankets, and so be prevented from getting into the front of the mic. What pickup was occurring from behind the mic could be further suppressed by using a small foam screen (around a foot square), or even a pillow, suspended directly behind the mic. Tony had been using a pop shield, so popping wasn't evident on any of his recordings.

Getting back to his existing recordings, Paul suggested that we try some different reverb settings to see if we could achieve a sense of space and air without the reverb getting in the way of the performance. One of the first things Paul has learned about *Logic's Space Designer* convolution reverb is to try even the unlikely sounding presets, as many of them sound quite different to way you expect them to. After a few trials, we eventually settled on Oak Tree Forest Short, a preset based on an impulse response recorded in a real forest, and we found that this gave the vocal a lovely sense of air, without the splash and mess of a typical artificial reverb. The vocal now sat much better in the mix, but we still needed to use compression to keep it even and make it sound more intimate.



Here you can see the settings used to process the lead vocal. The Oak Forest reverb patch was adapted to get the vocal to sit in the mix, while the compressor made the sound more even and intimate. EQ was then dialled in to emphasise breathiness and reduce an undesirable mid-range coloration.

Tony had been using a time-limited evaluation copy of the Waves *Renaissance Compressor* to treat the vocals, so Paul tried to set up *Logic*'s own channel compressor to keep the same character, but at the same time to apply more assertive control. The settings we arrived at were not atypical, and comprised setting the side-chain to RMS sensing (Peak is better from drums and other percussive sounds), with a ratio of 5:1 and a fairly hard knee. The threshold was then turned down until the gain reduction meter showed around 6-8dB of gain reduction on the signal peaks, but the exact degree of compression had to be set by ear. A fast attack time and medium release time seemed to work well, and Tony was very happy with the character of the sound.

The final step was to use gentle EQ to polish the sound, with the now-familiar broad-band boost applied at 10-12kHz to emphasise the breathiness and intimacy of the voice, while a narrow cut at 640Hz took out a slight hint of honkiness which could have been due to the room acoustics influencing the recorded sound. After re-balancing the vocals in the mix, we all felt they sounded more 'produced' and better integrated, without seeming over-processed.

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Bass & Drum Tweaks

A bass-guitar part had been treated using a rotary-speaker plug-in, but we felt this diluted the impact of the sound and prevented the bass from doing its job of driving the track along. The solution we tried was to remove the rotary effect and compress at a ratio of 5.4:1 using a hard-knee setting coupled with just enough compressor attack to let the start of the note come through cleanly. Again the threshold was adjusted so that quieter notes only just tickled the gain-reduction meter, while louder notes were subjected to a maximum gain reduction of about 6dB.

Finally, we used EQ to boost the 'character' part of the bass guitar spectrum, which in this case required a peak at around 230Hz and another at 500Hz. The normal instinct when treating a bass sound is to boost the bass, but this can be counterproductive, as most deep bass is inaudible on a small music system or car radio. By boosting the level in the 200-250Hz range, the bass sounds louder on small systems, and the emphasised harmonic frequencies help the bass remain audible when the other tracks are added to the mix.

A brushed drum part was processed to thin it out, so that it would remain plainly audible in the mix but not take up too much space. This was achieved mainly through the use of EQ, where a low-shelving filter was used to remove the bass and lower-mid-range frequencies, after which a 4kHz boost helped bring out the detail of the sound. The *Logic Exciter* plug-in is also very effective in this role if the original sound is inherently lacking in brightness.

Another track featured a nylon-strung classical guitar playing a rhythm part which

Tony had then tried to EQ to make it sound much brighter, almost like a steel-strung guitar. The obvious approach here would be to try recording the track again using a steel-strung acoustic guitar, as far less processing would then be needed to make it sound the way Tony wanted it to sound in the mix. Having pointed this out, Tony agreed at once that maybe a steel-strung guitar would work better in the song.

Tony then went on to say that he'd had noise problems when recording the acoustic guitar with a microphone, and had resorted to using a denoising plug-in to fix the track before mixing. Tony had been using a fairly high-sensitivity capacitor mic to record the guitar and vocals (AKG C414B ULS) in conjunction with a Drawmer 1960 tube preamp/compressor. There shouldn't have been any problem with equipment of this quality, so we asked him to demonstrate his recording technique with B playing a nylon-strung guitar. Sure enough, there was far more background hiss than there should have been. In checking the signal path it turned out that Tony was routing his 1960 into his Mackie mixer before sending it to the MOTU 828. The disparity in levels at various points along the way meant the overall gain structure was far from optimum — hence the noise.



This is the vocal recording setup Tony had been using. Paul & Hugh recommended that he face the mic back into the padded corner, rather than into the room, in order to reduce the amount of room sound captured.

Since all of the outboard gear was accessible via a small patchbay, it was very easy to patch the output of the 1960 directly into one of the MOTU 828 line inputs, and when we repeated the recording, the noise problem vanished completely. These little observations were appreciated by Tony as, no matter how much experience you have, bringing in a new set of ears (and eyes) always picks up something you hadn't thought of or had overlooked.

As a new pot of coffee was brought up, Tony asked whether we thought recording the guitars downstairs in a room with wooden floors would be better. The downstairs room sounded ideal for this, as the acoustics were nicely live (wooden floor, plain walls, and a very high ceiling), without being overbearing. An alternative solution would be to put a sheet of MDF or hardboard over the carpet upstairs. Recording instruments in another room two floors below Tony's studio would mean running long cables, but that's not really a problem. Arranging communications and headphone monitoring would be more of a headache, but it would be perfectly possible.

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Tony's Comments

"By the time of the visit, I was already enjoying much-improved monitoring having implemented some of Paul and Hugh's prior emailed suggestions. Moving the speakers a couple of feet away from the roof perling to their rear, and positioning them on solid concrete paving slabs (as opposed to a wooden shelf) atop concrete speaker stands had tightened things up considerably. The vast improvement in stereo imaging on the PMC TB1s (my main monitors) achieved by swapping their position with the NS10s was a revelation, as was the result of (ahem!) wiring the latter in phase.

"Paul had suggested making a wooden fabric-covered frame to house the Auralex panels, but as the roof of my studio is clad in 30mm-thick insulated plasterboard, I decided on a lightweight option which would have the desired effect, as well as making the panels easy to reposition if required. First I attached the foam to thin hardboard using spray adhesive, and then used a staple gun to cover the whole panel in light-coloured fabric. By cutting small strips of fabric away at the corners of the panels, I was able to drill screw-holes to fix the panels in place, which could then be covered up by tucking the fabric strips back under the edge of the panel.

"With the Auralex panels in place and bags of clothes and duvets behind the sofa to enhance its function as a bass-trap, the only further improvement to be made is the fitting of venetian blinds to the large Velux windows to further reduce reflections — and maybe I'll also upgrade to TB2s! I always enjoy learning from someone else's approach to a mix, and gleaned many useful hints and tips from watching Paul attack one of our tracks — particularly when it came to thinking 'outside the box' with regard to reverb selection. The decision to produce and release B's debut album without the assistance of a record label has many potential pitfalls, and we'd like to thank Paul and Hugh for helping to steer us in the right direction (their mastering advice has probably saved us a major headache). Over the course of the visit I found myself furiously making notes in a bid to remember as much as possible — the amount of information imparted in such a short space of time was a more than generous return for the coffee and Hobnobs!"



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Home Mastering

Inevitably the conversation turned to mastering and Tony mentioned that he had

resurrected his Tascam TSR8 half-inch eight-track recorder by having it fully serviced and the heads re-lapped. He'd originally thought of using this to record drum tracks to 'warm up' the sound, and also to use for location recording. However, it also occurred to him that he could use it while mastering to warm up tracks by recording the stereo mix to analogue tape, then playing it back into the computer. A number of engineers do this routinely, but Tony wasn't sure whether he should use just two tracks, or record the left channel onto the first four tracks and the right channel onto the other four tracks — he'd heard that doing this produced a quieter recording. While that is perfectly true — as the more tape area per second you can record onto, the lower the tape noise will be — even the smallest azimuth error in the mechanical head alignment would compromise the frequency response, so it's probably not a good idea to do this in practice.

The other point to keep in mind is the effect of the built-in Dbx noise reduction used in these machines. The Dbx noise reduction system only works well if you don't drive the recording levels into the red, since it relies on a clean recording to decode accurately. However, most people want to drive analogue tape fairly hard to get the soft-saturation warmth. If you're using the tape to create warmth, then running the machine at its highest speed and recording hot without the noise reduction switched on is the best bet. Because the recording levels are high, tape noise should not be an issue unless your music includes lots of very quiet passages, but of course the ultimate quality of your end result when working this way also depends on how good the converters are in your system, as you have to convert from digital to analogue and then back to digital again for the final mastering. Using a tape machine as a processor is certainly worth trying, but you should also compare the results with tape- or tube-emulation plug-ins, as the best of these probably sound better than an indifferent tape machine, and you won't be at the mercy of your converters. The level of noise is also likely to be much better using plug-ins.

However — and this is very important — you really shouldn't be applying any overall processing at all to your mixes if you intend to have them professionally mastered, and for this album project Tony was planning to use the services of a professional mastering house. The safest way to proceed is to mix your tracks as 24-bit audio files and then burn these to CD-R as data (WAV or AIFF files), not as audio CD tracks. Don't do fade-ins or fade-outs while mixing, as the mastering



The impact of a bass sound was increased using compression, the attack time being adjusted to let the start of each note come through cleanly. Some EQ was also used to increase the sense of body and character.

house can do them better and at the right point in the process, and don't top and tail the tracks, as the mastering engineer may need to derive a noise fingerprint from just before the start or just after the end of the recording if noise removal processing is deemed necessary at the mastering stage.

Of course if you are doing your own mastering, then you'll need at the very minimum a good compressor (either multi-band or full-band), a very nice-sounding equaliser, and a fast limiter. These can be hardware or software, but working in *Logic* there are good results to be had using the built-in compressors and channel EQ, although Paul has never really been impressed by the limiter — he normally uses Waves *L1* or *L2* limiter plug-ins. Mastering at home requires good ears and good monitors, as well as a knowledge of gain structure, the implications of the various digital recording sample rates and bit depths, and dithering. However, for privately produced records where professional mastering can't be justified, it can be done. What's more, you can learn a lot doing it!

By the end of the afternoon, we'd come up with some strategies for improving Tony's already good mixes and we'd also made some practical suggestions regarding recording techniques. But what we really wanted to know was where he'd got the wonderful sign: Hippies use side door! **SOS**

Published in SOS November 2004

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Studio SOS

Rob O'Neil

Published in SOS December 2004



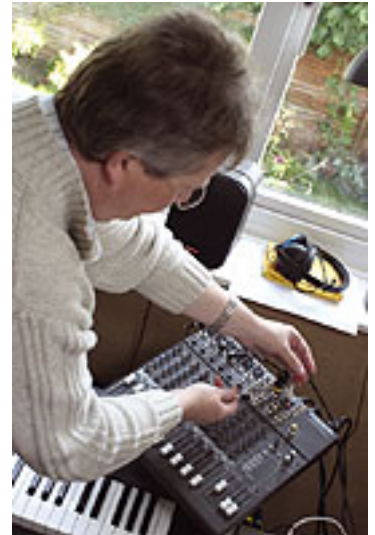
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People : Studio SOS

A dodgy snare recording was just one of the problems facing Rob O'Neil at the mixdown stage, so the SOS team visited his home studio to help him sort things out.

Paul White

Rob O'Neil is both a solo performer and a front man with his band Taxi, which comprises Matt Woosey on guitar, James Elliot Williams on drums, and Rob himself on bass and vocals. When Rob plays solo, the acoustic guitar is normally his weapon of choice, and when recording with the band, Rob's brother Bruce occasionally furnishes some fine piano parts while Rob overdubs acoustic guitar. They're a good band, with some great original songs, and they've supported major artists and played at serious venues, including the university tour circuit, The Rock Garden, and the Glastonbury Festival. When not playing, Rob has been known to do acting work and has turned up as an extra in several episodes of Casualty. Given his band's line-up and his traditional songwriter's approach to music, it's no surprise that Rob uses his computer-based studio more like a traditional 'audio only' setup.



Rob had been routing the output from his computer through a channel pair of his small Mackie mixer in order to monitor the signal through the mixer's main outputs. However, this approach put more electronics into the signal path than necessary, and also made it more difficult to use other channels for recording. The solution was to re-patch the computer's outputs into the Mackie's two-

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Assessing The Mixes

When we arrived at Rob's Worcestershire-based studio, he explained that he'd been on a bit of a

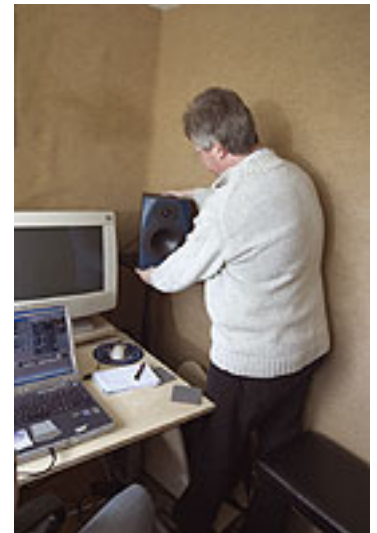
fitness regime so we were offered plain digestive biscuits rather than chocolate ones. I mention this only to ensure that you don't take this as some kind of precedent! Rob has what might be considered a typical bedroom studio, based around a Windows PC running Steinberg *Cubase SL* and augmented by a couple of synth modules and a keyboard. His main PC system had developed a fault when we visited, so he'd transferred everything over to his laptop, which was coping fine. He has one main vocal mic, a Rode NT1A cardioid capacitor model, and for monitoring he has Tannoy Reveal Actives set up on stands. While Rob enjoys producing his own material, recording a band with a full drum kit in a bedroom studio is not a practical option, so for his last project he used a small commercial studio to record the electric guitar, rhythm section, and vocals as a live take, then brought the WAV files back to his studio where he overdubbed the remaining parts before mixing them.

track tape return, feeding the monitors from the mixer's control-room outputs instead.

Although the performances were good and the songs well arranged, the mixes sounded somewhat congested and lacking in detail, which Rob first put down to the tracks needing mastering. However, once we arrived at his studio, it turned out that we could make more of an improvement by helping with the mixing. As always, the first job was to listen to a commercial recording played over the studio monitors to see how the room behaved. The room was 'L'-shaped and fairly small, and to help deaden the sound Rob had stuck carpet to most of the surfaces. Normally this isn't a good idea, as carpet only absorbs high frequencies, which can leave the room sounding quite dull or boxy. In this case, however, the presence of some plasterboard walls (which acted as *ad hoc* panel traps), a large window area, and the studio furniture helped to negate the potential ill effects of the treatment, and the general sound was quite workable.

However, the speakers were placed to fire straight down the room, and sitting in front of the computer screens to mix put the listener so far off axis from each speaker that the stereo imaging was very poor and unstable. All we had to do, though, was turn the speakers in slightly to face the mixing chair and the imaging improved enormously. We also noticed a slightly bass-light area in the centre of the room. Although proper bass trapping may help to improve this, it wasn't a serious problem as long as we avoided sitting in the middle of the room while mixing!

Rob was using a Mackie 1402VLZ mixer for monitoring, where the output from *Cubase* was being fed into two mixer channels panned left and right. His monitors were then fed from the main stereo outs. We suggested that he instead run the output from *Cubase* into the two-track returns on the mixer and then select those inputs as the monitoring source. Of course his speakers would



The stereo imaging was being compromised because Rob hadn't angled his speakers in towards the monitoring position, so Hugh quickly remedied this.

need to be connected to the control-room monitoring outputs on the Mackie mixer rather than to the main outputs, but this only meant changing the XLR cables for balanced jacks. The advantage of this setup is that the main mixer channels, via the main stereo out, become available for recording into Cubase at the same time. Furthermore, as the two-track return has no EQ, the simpler signal path should make for more accurate monitoring.

First off, Rob played us a mix of one of his songs, and this did indeed sound a bit middly and congested, though the balance was pretty good. Rob said that his method of mixing was to treat each sound in isolation until it sounded as good as possible, then balance the sounds in *Cubase*. However, as you may have discovered if you've tried it, this approach doesn't always work as well as you might expect. The important thing is how the sounds work against each other in the mix, not how they sound in isolation. When checking individual sources prior to mixing, it is worthwhile correcting any obvious EQ problems, but you should leave the creative tonal adjustments until a rough mix is achieved, as you're almost certainly going to need to make changes then.

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Salvaging A Drum Recording

Rob's job was further compounded by the drum sound the studio had achieved. They had used a conventional pair of overheads augmented by a kick mic and a further mic placed near to the snare and hi-hat. While the overhead mics sounded OK, the snare-drum mic was producing an extremely boxy sound with virtually no top end, almost as if the mic had been facing the wrong way. I don't know what had gone wrong in the original recording — although the snare mic sounded as dull as ditch water, the snare sound in the overheads was bright and natural. The close-miked kick drum was also ringing and lacking in definition, with quite a lot of spill, and it sounded as though the mic was placed in front of the kick drum rather than inside.

Sorting out the drum sound was clearly going to be our greatest challenge, as the other tracks sounded perfectly usable, including the overdubs Rob had done at home. Had we had a decent enhancer plug-in, I would have used that to add some snap back into the snare drum, but I ended up using a 115Hz cut combined with a fairly aggressive 6.3kHz boost. I then gated the track to clean up the spill from the other drums, and to lose the ring. This was all done using the *Cubase* plug-ins, as they were pretty much the only ones Rob had. In isolation, the sound still wasn't very bright, and the gating



effect wasn't entirely natural, but checking it in combination with the overhead mics showed that the composite sound wasn't that bad. There was a reasonable amount of snare 'snap' in the overheads anyway, and the tweaked close mic helped add in some welcome body and weight.

The drums were left mainly dry, but a little *Cubase* 'Ice' reverb — a nice bright plate-like sound — was added to the snare track, just enough to give the impression of playing in a larger space, but not enough to wash out the sound. For the kick drum, we settled for a

250Hz cut to try to take out some of the boxiness, coupled with a 5.5kHz boost to try to improve the definition. Again gating was used to take out the ring. Problems can arise when mixing the overhead mics and the close mics together, because small phase differences caused by the spacing of the mics can compromise the low end. We solved this simply by rolling off some low end from the overhead mics, adding a little gentle air EQ at 12kHz or so, and relying on the two close mics to add the weight.

The two overhead mics were presented as separate mono tracks, rather than as a stereo pair in *Cubase*, so we copied the EQ settings from one channel to the other to ensure they remained properly matched. Whereas Rob had balanced the close-miked sounds and then added a little overhead to fill in the cymbals, we worked the other way round using the overheads as the main sound and the close mics just to underpin the kick and snare. To our ears, this gave a much more open and transparent drum sound. Finally, the amount of snare reverb was adjusted to give the right subjective reverb level when the rest of the drums were playing.



Although the overhead drum mics sounded good, the snare and kick close-mic signals had been badly recorded, and needed some heavy processing to salvage something usable — here you can see the settings used.

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Mixing The Overdubs

The next thing to add was the bass guitar, which had been DI'd directly into the desk. In isolation, this sounded fine, but tended to lack definition when the remaining tracks were brought up. This is very common with DI'd bass, and the instinctive response is to add more low EQ to the sound to bring out the bass, but that tends to be counterproductive as it means you have to turn the overall track level down to maintain headroom. The part of the bass-guitar sound that you actually hear in a mix is more likely to be between 200 and 300Hz (which is why you can still hear the bass guitar on a small transistor radio with no low end) and that's the area that gets reinforced when you use a bass guitar amp and speaker rather than DI'ing. Our solution was to compress the bass slightly to even out the level, then boost it at 255Hz to give the sound the ability to cut through. Rob

wasn't sure about this when he heard the bass track in isolation, but with the rest of the mix up and playing, he agreed that it worked much better.

Rob's acoustic guitar was one of the other mainstays of the mix, but he'd again EQ'd this in isolation, with the result that it sounded rather like a solo acoustic guitar part. When used within a mix, the acoustic guitar doesn't need that big low end — it just gets in the way — so again we added mild compression to keep the level even and applied a fairly wide cut at 250Hz to take some of the weight out of the low end. There was also a hint of harshness in the sound which was dealt with by cutting at around 3kHz. This left us with a thinner sound with more zing to it, which sat well in the mix without getting in the way of the bass guitar and kick drum.

Rob's lead vocal was pretty good already, so that was treated with just very light compression and a touch of 12kHz 'air' EQ and light reverb to freshen it up. However, when vocals are going to be recorded in a bedroom studio such as his, the result can generally be improved by using our regular trick of hanging a duvet or sleeping bag behind the singer's head to reduce the amount of reflected sound getting back into the mic. In practical terms, this reduces the effect of the room acoustic on the sound, and even though a little room coloration may sound OK to start with, as soon as you add compression it tends to get emphasised. The less room sound you pick up, the cleaner the vocal recording will be.

As a rule, it's best to try to avoid radical EQ with vocals, as it can lead to a very unnatural or coloured sound, especially with simple plug-in equalisers. It also helps to be sparing with less sophisticated reverb plug-ins, as they have a habit of seeping into all the spaces in a mix if you use too much. This is often not fully appreciated unless you've had a chance to hear a really good reverb, so upgrading to either a mid-price hardware reverb or a DSP-powered effects card/box is a good way to upgrade if your system doesn't have the horsepower to run



After working on the mix, Paul worked with Rob on selecting suitable mastering processing — the final settings can be seen in the screenshots.

one of the new convolution-based reverb plug-ins.

An existing electric guitar part was left much as it was, but re-balanced lower in the mix, and the guitar solo — which was on a separate track — was tweaked to make it sound more part of the band and less like an overdub. Rob had added some big delays of around three and four seconds respectively to the left and right channels, as well as some fairly dramatic EQ, but the guitar sound had a slightly rough edge to it and the big delays made it sound quite disembodied. As the flavour of this song was more 'West Coast rock', something more conservative was required. The guitar tone was tamed by applying a 92Hz cut to thin out the bottom and a 6.9kHz cut to smooth out the top. The left and right delays were reduced to around 300ms and 400ms, resulting in a sweeter, better-integrated guitar tone.

Last of all, a piano track, played by Rob's brother Bruce, had a strong left-hand bass component that tended to fight with the bass guitar and kick drum, so some low cut was applied to keep it out of the way. Then we balanced up the tracks and retired to the landing outside the bedroom studio to listen to the result from outside the door. This trick of listening outside in the passageway never fails to show up balance problems, after which we made a few further minor level adjustments and then bounced the song to a stereo file that we could use as a 'guinea pig' for some mastering experiments. We also did some straight comparisons with Rob's original mix, which showed we had improved the overall transparency and clarity of the mix, even though the snare drum sound was never going to be great.

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Rob's Comments

"Having the SOS team visit was extremely useful in several respects. Firstly, the band and I didn't listen back to the original studio recordings in enough detail, just a rough monitor mix, so we didn't pick up on the fact that the kick and snare sounds were extremely poor — as Hugh and Paul pointed out, they were being concealed by the overheads, so no-one noticed at the time. We were aiming to capture a performance, which took precedence over sonic quality, though with hindsight we should have had both in mind equally. However, the budget didn't allow for too many takes — a maximum of two shots at each song!

"Observing the remix was extremely informative. I had approached it wrong by trying to get each individual sound as good as possible in isolation first. I now see the process more like a jigsaw — the pieces don't fit if you have segments that overlap. Removing elements from each of the sounds really gave the mix room to breath, as though you could see daylight between the instruments. I thought it sounded more 'live' than my mix,



more real. I'd tried to use low frequencies and volume to give it power, but had achieved the opposite!

"One great tip was how to make the lead guitar solos sound bigger. Having applied big delays to try giving it that stadium feel, Paul & Hugh made it more subtle, explaining that you can get the same effect of making it sound like it's in a bigger space just by how you EQ it and place it in the mix. Overall the power didn't suffer for reducing the bottom end on things like the bass guitar. The whole mix became clearer, sweeter, and more punchy; the guitars weren't fighting for space; and the vocal sounded really full and louder, but still part of the band.

"Once the mix was lighter and more free of overlapping sounds, balancing the track was much easier, and all the instruments seemed to slot into place where I wanted them to do so, without any persuasion required. From now on I'll work the other way round, do a rough mix first, then start throwing out elements that are occupying the same frequency band before mixing again. It all goes to show how much skill and experience is needed when mixing — I've always concentrated on writing and performing, and engineering is equally difficult to master. I'll apply the techniques the team showed me and get more engineering practice done!"

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Mastering Suggestions

Rob asked us if it was OK to use mastering processors at the mixing stage, and while there's no technical reason not to do this it doesn't really help achieve one of the main aims of mastering, which is to help the different songs on an album work together both tonally and in terms of level. It's far better to mix with no overall processing, then compare the finished tracks that will be used on an album before processing them to produce the same 'family' sound. Other considerations such as adjusting relative balance, and limiting to achieve extra loudness, can also be addressed at this stage. Personally, I wouldn't try to master anything serious in a small bedroom studio using nearfield monitors, but if you check your mixes on other systems, you can achieve some worthwhile results and gain experience at the same time.

Neither are the basic plug-ins that come with 'light' sequencers particularly great sounding for mastering, but used with care they can help apply a little polish to mixes. On this particular mix, we demonstrated the gentle compression that can be achieved by using a very low compression ratio of around 1.2:1 combined with a low threshold (around -35dB relative to digital full scale) to produce just a few decibels of gain reduction across the entire dynamic range of the mix. This was followed up by the *Cubase* limiter and soft clipper, and the compressor makeup gain was adjusted to give brief limiting on signal peaks. I didn't feel the *Cubase* limiter had good enough metering to do this job in any serious capacity, but we managed to get a lot more subjective level without obviously compromising the sound. There is no standard EQ for mastering, as you just have to do what the song needs, but we demonstrated the clarifying effect of broadband 'air' EQ in the 12-14kHz range as well as the ability to add 'thump' by applying just a little boost at between 80Hz and 90Hz.

Although Rob had made good recordings and clearly has a good ear for balance,

his mixes had run into trouble because he was trying to optimise sounds in isolation before mixing them. The studio drum sounds he had to work with didn't help either. While you can make general adjustments before mixing, you are nearly always going to have to change your EQ settings once everything else is playing. The other thing that helped us improve the mixes was using EQ to prevent sounds colliding with each other, especially in the upper reaches of the bass register, where things can easily get messy. This kind of attention results in a much more open-sounding mix with far greater clarity — as Rob himself commented. Rob had also been using some over-dramatic EQ and compression treatments to shape his individual tracks, but readjusting the EQ settings made a big difference, and most of our more 'dramatic' EQ settings involved cut rather than boost, as the ears perceive that as more natural. Rob also tried too hard to push the guitar solo out front by making it loud with lots of echo. By re-equalising it and adding a more restrained amount of echo, then pulling it back in the mix a bit, it sounded as though it belonged to the band again and didn't sound any smaller or less powerful for it. **SOS**

Published in SOS December 2004

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Studio SOS

Jiang Li

Published in SOS May 2004

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People : Studio SOS

The SOS team apply themselves to the task of mixing Chinese traditional instruments with mainstream Western sounds at reader Jiang Li's home studio.

Paul White

Jiang Li not only records himself and other musicians at his small home studio, but also collaborates with musicians from different parts of the world over the Internet using a system similar to that pioneered by Rocket Networks. However, he was having a few problems with his recording and mixing and so invited us to his studio, which is packed into a fairly small ground-floor room in his Northamptonshire home.

The studio is a little under 3.5m wide and about 2.5m deep, with a ceiling height that is typical of modern houses.

The entrance door is in the rear wall of the studio and, once all the gear is in place, the engineer's seat is well under one metre from the door. The monitoring is courtesy of a pair of Mackie HR824s set up on a rigid shelf above the Behringer Eurodesk mixing console.

His latest acquisition was a Mac G5 running *Pro Tools LE* using a Digi 002 interface, but along with many other G5 owners Li had discovered an unacceptable rhythmic background noise problem when using a Firewire audio interface. This isn't something we could deal with, as it seems like an Apple hardware-related problem affecting some G5 machines — for more details about this, check out Mark Wherry's Apple Notes column this month. Because of this



A little Auralex foam put paid to some troublesome wall reflections in Li's room, but Hugh also felt that the speakers were not angled optimally for a seated engineer, so he wedged them up at a slight angle in order to aim the tweeters more appropriately.

noise problem, Li has kept his *Cubase* project on his older 450MHz Mac G4, which works perfectly but tends to run up against its processing limits when lots of plug-ins are being used. He also has a fast PC laptop (with an external 200GB Maxtor drive) which is used to run Steinberg's *Wavelab* for editing, and also Tascam *Gigastudio*.

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Quick Mastering Tips

It's always a concern when clients complain their mixes sound quiet next to commercial recordings, but the truth is that many records end up being overcompressed purely to meet this need for loudness. Li was interested to know what treatment was appropriate, given that much of his material would not be commercially mastered. So we suggested cautious post-EQ limiting (just a few decibels off the loudest peaks) and gentle EQ to clarify the mid-range and to bring out high-end detail. Using an analogue EQ, this usually means applying around 2dB of cut in the 180-200Hz region and adding a wide, gentle boost of no more than a couple of decibels at 12-14kHz with a Q value of not higher than 0.7, though this has to be done while listening to see what the track needs — it shouldn't be used as a blanket sweetener for everything.

Like most digital equalisers, I found that I had to apply more cut and boost with the *Pro Tools LE* parametric EQ plug-in to get the same subjective result, but operating the bypass button showed that the difference was suitably subtle. Again, comparing our processed result with Li's original mix showed that our version sounded noticeably louder and it had more clarity, so these simple tricks clearly work, even though there's a huge amount more to 'real' mastering than this.

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Noise Remedies & Monitoring Tweaks

There was a noticeable hum on the system at high monitoring levels, but because of the way everything was packed into the room we couldn't do any impromptu rewiring. We did however suggest that Li tidy up his mains connections and route them all from the same wall socket, because the overall power requirement was low enough for this not to be a problem. Further improvements can be made when connecting unbalanced sources, such as synths and samplers, to a mixer with balanced line inputs by using the unbalanced-to-balanced cable connection system whereby the balanced end of the cable is wired as normal (using twin-cored screened cable), but at the unbalanced end the screen is left disconnected and the cold (usually black or blue) cable core is connected to the tag where the screen normally goes.



The Digi 002 rackmount interface went dead during

Before listening to Li's mixes in detail, we wanted to evaluate the monitor system's performance, so we auditioned some commercial mixes through the system and felt the result was rather bass heavy. Checking out the switch settings on the rear of the HR824s revealed that Li had set them to the correct half-space mode, but with their full bass extension. I found this setting was too low for my own studio, which is more than twice the size of Li's. I have my bass switches set to their middle positions (47Hz), so suggested Li tried the same. The other issue was that the shelf supporting the monitors was quite high, resulting in the tweeters being well above ear level when Li sat in the chair to mix. Therefore we placed a wedge under the rear of each speaker so that the tweeters were pointing toward where the engineer's head would be when seated.

the session, but started up fine again after a lunch break. Hugh suspected that it might have been overheating, so he partially removed it from the rack to increase ventilation for the remainder of the session.

Repeating the CD playing tests indicated that the balance was now more manageable and, surprisingly, the bass end was fairly even, probably due to the thin rear door acting as a bass trap (or bass leak) and the window in the side wall also tending to do the same. However, the room was not acoustically symmetrical, as the window was to the right and fitted with slatted blinds (helping to break up reflections) while the left-hand wall was bare painted plaster. To help even this up, we brought in some Auralex foam oddments left over from a previous Studio SOS visit and leant one up against the left-hand wall, put one behind the monitors, and propped one up against the hard surface of the door. We left it to Li to find a way to fix these more securely without actually gluing them to the walls — it's easy enough to fix a piece of string across the back and hang them up on picture hooks. A re-test with the commercial CDs showed that the image had clarified to a worthwhile extent, and at this point we felt we had got the monitoring about as right as we could, given the physical constraints of the studio.

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Processing Submixes

Next we turned our attention to a mix that was typical of the ones Li was having problems with. This one comprised a mixture of western and Chinese instruments, where the orchestral sounds and some of the percussion sounds were samples, the acoustic bass sound came from a Proteus 2000, and the voice and acoustic Chinese instruments/percussion had been recorded by Li. The problem Hugh and I identified straight away was that the recorded acoustic instruments didn't sit very well alongside the samples — the sampled orchestral sounds had a concert-hall feel, while the acoustic instruments sounded as though they were still in a fairly small room, even though Li had added



some reverb.

Once the monitoring system was in order, Paul set about tweaking Li's mix to get the sampled instruments sitting alongside the recorded ones.

The vocal also sounded a little thin and uneven in level, and where Li had submixed a number of Chinese instruments to a single stereo track we found that some of the instruments really jumped out while others were getting buried. It seems that the limitations of the G4 had forced Li into working with stereo submixes of sets of the original tracks. It would have been preferable and more flexible had he kept the individual instruments separate and controlled them with fader groups instead. However, this is something he can explore at leisure using his G5 once he's sorted out the noise problems.

To make the best of the existing job, we first compressed the vocal track using Li's Focusrite Penta compressor, which has presets for various applications. Although I don't normally like using presets, the Penta allows further changes to be made once a preset has been selected, so we used the Vocal preset as the obvious starting point and then adjusted the threshold to give five or six decibels of gain reduction on the loudest peaks — which seemed enough to even out the sound. We also added just a small amount of a Chamber reverb patch from Li's Lexicon Alex processor to make the vocal sit in the same kind of space as the sampled backing parts. Good reverb is very important when mixing, but the Lexicon Alex is now rather long in the tooth and probably isn't much of an improvement over run-of-the-mill native reverb plug-ins. Once Li's G5 is behaving properly, he'll have power to spare to run a convolution-based reverb plug-in (such as Audio Ease *Altverb* or the new Waves plug-in), which should make a vast improvement in this area.

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Global Musical Collaboration Network

Jiang Li has been bringing music from the East and the West together for the past ten years as a musician and composer, but he has also recently been awarded a fellowship programme from NESTA (National Endowment for Science, Technology and the Arts) to develop the web-based Global Musical Collaboration Network. The idea of this is both to educate people about music from different cultures (via audio and video examples of music from different cultures) and to facilitate co-operative projects by allowing musicians from different parts of the world to collaborate. Although this free-membership community is still in its infancy it already includes a live chat room, a forum, a technical FAQ section, collaboration projects, and an on-line radio station. It also includes a showcase section for members to publish themselves and to search for collaborators worldwide. As a registered member, you can take part in projects and also start up your own. The different levels of collaborations are suitable for amateur musicians up to professional studio audio engineers.

 www.globalmusicalcollaboration.net

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Rebalancing The Mix

That left us with the Chinese instruments mix to sort out, but compressing this with the dynamics section in *Cubase* happily made a huge difference. Given the levels Li had recorded, the threshold was set to -20dB and the ratio to 5:1, with the attack time fast and the release time on automatic. These settings worked well to control the loud instruments and to lift out the parts that were previously getting lost.

Before rebalancing the track, we thinned out the string samples by cutting back the low end, and also took off a little top, as they were just starting to sound strident. A further supporting pad was brought down in level. We tried to EQ the string bass sound to get more definition, but found that we needed huge amounts of EQ boost to do this, as there was very little presence in the original sample. The problem with using this amount of EQ was that the string squeak sound Li had added really leapt out, so we used a compromise setting and suggested that Li find some good acoustic bass samples to use for his next project, rather than trying to re-shape the bass sound from his *Proteus 2000*.

To mix the track, we started with all the faders down and then balanced the bass, drums/percussion, and vocal so they worked together. After that it was fairly straightforward to bring in the strings and brass samples, the Chinese instrument mix, and the individual Chinese instruments to achieve a cohesive balance. Where a little reverb was needed to add a sense of space, the same Lexicon Alex Chamber preset was used. When we compared this remix with Li's original version, it transpired that our mix had a higher level of drums, but what was most noticeable was that the individually recorded instruments no longer sounded out of place, and the overall feel of the mix was less crowded because we'd allowed some of the parts, specifically the strings and pad, to play more of a supporting role.



The bright ambience of Li's kitchen was ideal for recording Li's traditional yangqin.

Li asked if we had any special techniques that could be used to create more of a sense of front-to-back perspective, and in some ways we'd already applied these when mixing. As a rule, upfront sounds tend to be less reverberant and brighter than those at the rear, so if you want to keep a vocal at the front, use a shorter, brighter reverb or ambience program, and perhaps use EQ to add a little 'air' into the sound above 10-12kHz using a wide parametric boost. Sounds to the rear can afford to lose a little high end and can be made more reverberant. On this particular track, the vocals were more of a supporting effect than a lead line, so we could afford to use more reverb.

At around this point, the Digi 002 decided to die on us, and after trying to reboot it and checking the connections, we gave up and took a break for lunch —

traditional Chinese *dim sum* presented by Li's wife. It made a welcome change from chocolate biscuits, but don't let that put anyone off offering them on future Studio SOS projects — after all, we have a tradition to uphold!

After lunch, we re-powered the Digi 002 and it sprang back into life, so we wondered if its earlier failure was heat related, even though it was placed at the bottom of the open rack below the Penta, and wasn't exposed to any obvious major heat source. Just in case heat was the problem, though, Hugh temporarily removed the Digi 002 from the rack and we continued to use it this way for a while. However, it suffered a brief relapse later in the afternoon, so we felt it should be returned to base for a checkup, as something was clearly not quite right. Li has since reported that he has rearranged the rack and upgraded his Mac OS to 10.3.1 (which is approved by Digidesign) and the problem seems to have disappeared. Anyhow, it limped on gamely for the rest of our visit, allowing us to experiment with some overall 'mastering' processing. For this part of the process, Li moved our new mix of the song into *Pro Tools* so that we could use its plug-ins.

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Li's Comments On The Session

"I work with Chinese musicians a lot of the time, but have never been quite satisfied with my mixes. Also, some Western instrument sounds, such as bass, strings and so on, are difficult to integrate. Now, by adjusting a few settings on my existing gear, making some simple setup changes to the monitor speakers, and adding a little acoustic foam, I can already hear a great difference. Paul set up different EQs for different instruments on my Eurodesk, and a lot of unwanted frequencies disappeared, making the mix clearer. The compressor also made parts more audible without them being too loud.

"The Studio SOS visit made me more confident and positive, in particular when I'm working with Chinese musicians who generally know very little about the technology. Paul and Hugh also looked at other rooms in my house and discovered that they were suitable for different kinds of live recordings as well, which literally expanded my small studio to include the kitchen and living room. I can now put off my plans for renting a studio elsewhere and use some money for the next items on my gear shopping list, in particular a couple of mics similar to those that Paul and Hugh brought along."



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Recording Traditional Chinese Instruments

Once we had shown Li our approach to mixing, the conversation turned to the

recording of traditional Chinese instruments, of which Li has a fine collection. Although Li had a couple of decent mics, including a nice Audio-Technica 3035 and an ATM31A, he wanted to be able to record up to four musicians at a time and so was in the market for a couple more instrument mics that wouldn't break the budget. I had a couple of SE Electronics SE1s in the car, which provide a decent example of what you might expect from an entry-level small-diaphragm capacitor instrument mic.

Li had been recording instruments in his hallways, which sounded boxy to the ear and gave, I felt, a rather coloured and enclosed tone to some instruments, especially those that couldn't be miked very closely. As Li can use any room on the ground floor of the house for siting the instruments, we checked out the lounge and the kitchen to evaluate their suitability. The kitchen has a hard floor and lots of hard surfaces and so, predictably, sounded quite bright, which we felt could sound good on acoustic guitars and other stringed instruments.

To confirm this, we miked up a hammer dulcimer (called a yangqin) with a single SE1 mic via Li's Digi 002 preamps and got a very acceptable sound right away. For a 'real' recording however, I suggested that it would be a missed opportunity not to mic this instrument in stereo where possible, so whatever new mics Li bought, he should get two of the same so that he could use them as a stereo pair.



I've heard of playing 'Chopsticks', but this is ridiculous...

By contrast, the lounge was carpeted and contained a large three-piece suite covered in heavy fabric. The result was a much drier sound that would be ideal for vocal recording. We demonstrated this by me walking between rooms, holding a mic and reading a manual, while Li and Hugh listened (and sniggered!) in the control room. The change in sound when moving from the kitchen to the lounge was massive so, unbeknownst to him, Li already had two very different and very useful recording rooms at his disposal. Even the coloured-sounding hallway could be useful for some sounds, and where good isolation was needed three instruments could be recorded simultaneously in the three spaces, plus a fourth in the studio, with very little spill.

Each Studio SOS visit throws up some common themes and some unexpected questions. In this instance, I think we were able to show Li a slightly different approach to mixing and sound shaping, and exploring the acoustic spaces in his house was also a worthwhile exercise. However, all this is pointless if the monitoring system isn't giving a good account of the truth, which is why we always try to optimise this at the outset. This visit was also unique in that the word duvet was never mentioned! Damn, I've blown it now... **SOS**

Published in SOS May 2004

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Studio SOS

Alan Pittaway

Published in SOS January 2003

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People : Studio SOS

This month SOS helps Alan Pittaway improve his drum kit recordings, and also helps him get his two Roland multitrackers working together.

Mike Senior

Now that samplers and physical modelling processors are so affordable, it's possible to get commercial-quality recordings without miking up anything more than a vocalist. It's hardly surprising, then, that some home studio owners find the task of multi-miking a drum kit a bit of a challenge. So when Alan Pittaway emailed through to the SOS office saying that he was having trouble with his drum recordings, I decided to head down to his home studio in Buckinghamshire and help him out.

Alan is lucky enough to have a room in his house set aside as a studio, and he uses it to record his friends and various local musicians on a pair of Roland VS880 recorders — one an old VS880 VXpanded and the other a newer VS880EX. However, he'd not had much luck in getting the kinds of drum sounds he was after, even with a fairly decent selection of mics and outboard gear to hand.

Setting Up Overheads

Once the kettle had been boiled for the first cuppa of the day, the first thing to do was to have a listen to the kit while Alan manned the sticks. Although the studio room was quite large (about 7 x 4m), the carpeting, blinds and furnishings were doing a very good job of damping reflections, so the sound was quite tight. Although I'm not a drummer myself, I felt that the drums themselves sounded pretty good, so we pressed on with setting up mics. Alan's usual way of working thus far had been to set up the kick and snare close mics first, and only then to add in the sound of an overhead mic. However, I fel



t it would be best to start work with the overheads first, because of the lively sound he was after — he had suggested Blondie's recent single 'Maria' as a good reference, so we were treating that sound as our goal.

The overhead mic Alan had been using up to that point was a single boundary microphone dangling from the end of a mic stand on a piece of wire. This was presenting the drums completely in mono and was also messing with the mic's frequency response — boundary mics are designed to be placed against a large flat surface such as a wall, and their responses are significantly different without this assistance.

Fortunately, Alan had a pair of much nicer Rode NT1s, so I suggested we use these as overhead mics instead.

We ran them first through old Studiomaster console, to provide them with phantom power, and then straight into the VS880EX. I decided to use the newer Roland multitracker because it has slightly better converters, and also because it has phase inversion facilities missing on the earlier model.



The multi-mike setup used for recording Alan Pittaway's drum kit.

After setting up basic levels on the mixer and multitracker, I grabbed a pair of enclosed headphones to check the sound while I set up the mics. Positioning the mics a few feet above the kit on either side, angled downwards towards the cymbals, the sound seemed to be fairly well balanced, so I did a quick test recording. Listening back to this on Alan's B&W speakers, we had an acceptable unprocessed overhead sound, with the cymbals nicely balanced, so we decided to start adding in some close mics to give definition to the individual drums. Before we moved on, though, I asked Alan if he could furnish me with a couple of bricks to weigh down the legs of the boom stands — they were both quite extended, and I didn't want either one overbalancing while we were recording.

That Sync'ing Feeling

Like many owners of smaller multitrackers, Alan wanted to synchronise two recorders to increase the simultaneously available I/O and track count. As I described in the Q&A pages of SOS November 2002, combining digital multitrackers has its limitations, particularly in terms of combining aux buss signals from the two machines, but this didn't worry Alan in his situation so we set to work with hooking the two machines together.

The first task was to synchronise the two transports, so that the older VS880 controlled the newer VS880EX. Digging into the master unit's System menu first, we headed for the MIDI Prm submenu and set the MMC Mode to Master, making sure that MIDI Thru Sw was set to Out and SysEx Tx was set to On. Next we went into the Sync/Tempo submenu and set Sync Gen to MTC, ensuring that MTC Type was set to 30 and Source was set to Int. Moving over to the slave machine, we went to the System menu's MIDI Prm submenu and Set the MMC Mode to Slave, checking that SysEx Rx was On. Finally, we opened the Sync/Tempo submenu and set Source to Ext, making sure that MTC Type was set to 30.

That's an awful lot of parameter juggling, so here's how it all fits together. We set the master recorder to transmit two kinds of MIDI message: MIDI Time Code (MTC) and MIDI Machine Control (MMC). The MTC messages tell the slave where in the song the master is located, and the MMC commands operate the slave's transport controls remotely. The slave was set to recognise each of these types of MIDI message, in particular the 30fps MTC being sent by the master. With these settings made, the VS880EX responded to the location and transport operations of the VS880, and the playback of the two machines remained in sync once playback was started. Furthermore, with tracks enabled for recording on the slave machine, operating the master's Record transport button enabled recording on the slave machine, even when no tracks were record-enabled on the master.

At this point, however, there was only control information passing between the two recorders, so our next job was to get audio from the slave machine's mix buss into the master machine's mixer. A dedicated S/PDIF cable was used to connect from the digital output of the VS880EX to the digital input of the VS880, and then the Master Clk setting in the System Prm submenu of the VS880's System menu was set to Digital. The last step was to go into the VS880's master fader editing parameters and set Stereo In to Digital. Once this had been done, we pressed play on the VS880, and were rewarded with the sound of the VS880EX's mix combined with the VS880's.

However, there was one fly in the ointment. I noticed that the master meters on the VS880 were reading lower than those of the VS880EX when only the VS880EX was playing. I couldn't find any reason why this should be the case, so I switched to the VS880's Input Mix/Track Mix mode instead, and used input channels five and six (stereo linked and with the buss switch set to pre-fade) to feed the VS880EX's audio directly to the VS880's mix buss. This gave identical meter readings on the two machines, as I'd expected from my own experiences, so I can only assume that some level offset is automatically incurred by using the stereo input facility in Input To Track mode. I don't like the idea of a mixer doing things to the audio without it being told to, so I suggested to Alan that he get to grips with Input Mix/Track Mix mode to avoid this — anyway, it's much more flexible and also makes bouncing tracks and effects much less hassle.

Adding In Kick & Snare Close Mics

To start with, I set about adding in the close-miked sound from an Audio Technica DT25 which was already set up inside the kick drum. This was routed through Alan's Focusrite Penta preamp/compressor directly to the VS880EX. After the basic level-setting had been done, we did another quick test recording so that we could experiment with the phasing of the kick mic against the overheads using the VS880EX's internal phase switches — the mic combined best with the overheads without inversion, so we left the switch out on the preamp.

One of Alan's biggest concerns on contacting SOS was that his kick drum sound didn't have enough attack or weight, and this was apparent from the sound we were getting — although a couple of cushions inside the drum made the sound fairly tight, it lacked any real definition. Straightaway, I moved the mic closer to the point where the beater was hitting the skin, to get more of a 'click' at the

start of each hit, and then I had Alan patch in one channel of his Aphex 109 EQ between the Penta and the VS880EX. Although we could have used on

e of the multitracker's internal digital equalisers after recording, these have a rather harsh sound in my experience (especially when boosting), so I find that I get the best results by doing most of my EQ'ing in the analogue domain while recording. With Alan playing the kick, I used the Aphex to add about 4dB boost in a fairly narrow band at 80Hz for weight, and I also emphasised the beater click with a couple of decibels broadband boost at around 2kHz.

Another audition on the speakers confirmed that these changes had improved the sound considerably, but I felt that a little compression might also help emphasise the attack, so Alan headed back to the kit while I dialled some in. Starting from the Penta's Kick preset, I slowed down the default attack setting to let through the sound's attack, and then adjusted the Compression control until about 4dB of gain reduction was showing up. Once I'd slowed down the release a little as well, to avoid messing with the kick's decay, another test recording was done. Alan professed himself pleased with this sound, so we decided to move on to close-miking the snare.



An SM57 was positioned looking over the top lip of the snare, pointing towards the centre of the drum and away from the toms.

Alan already had an SM57 set up on the snare, although its positioning was a little out of the ordinary. It was pointing across the drum from the hi-hat side towards the tom-toms, and Alan told me that this was to get maximum separation between the snare and hi-hat. Given the sound we were after, I felt that greater separation between the snare and the tom-toms would be more important, so I repositioned the mic to a more usual position looking over the rear lip and angled slightly downwards towards the centre of the drum — the hi-hat was still about 90 degrees off axis. We routed the mic through Alan's Focusrite Voicemaster to the VS880EX, and set up sensible levels using the voice channel and recorder metering. Another quick recording confirmed that the snare mic sounded best with its phase flipped relative to the other mics, so we engaged the phase button on the Voicemaster. I also activated the Voicemaster's low-cut filter, set at 100Hz, to avoid spill on the mic interfering too much with the low end of the kick drum we'd already set up.

Listening back to the recording, we could hear the snare ringing a little too much, so we tackled the problem at source by gaffer-taping some rolled-up tissue paper at the edge of the batter head in two places. This sorted out the ringing nicely, but Alan still felt that the sound needed more 'sparkle' to it, so I hit the snare while he experimented with the EQ on the Voicemaster. In the end, we added about 4dB of Presence and 2dB of Breath before we were satisfied with the sound.

Alan's Session Notes

"When I asked SOS for help I was having two problems. The first was that I'd been unable to sync up my two Roland VS recorders, so I was limited to recording a maximum of six tracks at a time — I felt like I was driving an expensive car, bristling with features, but with little idea how to use them once the engine had been started. Mike arrived at about 10am (with some alarmingly expensive-looking boxes and mics — thoughts of selling Granny came to mind) and in less than an hour he had them talking to each other. Terrific! Now I can mike up a drum kit and still have tracks spare for guide vocals or a bass line. If you already own a VS880 or VS840, then getting hold of another one second-hand will greatly improve your options.

"The second problem was my supreme ability to make a top-notch drum kit sound like cardboard boxes being played in the deep end of the local swimming pool! At first I was confused that Mike spent so long setting up the overheads, moving them around and tweaking the sound. I normally sort out the snare and kick drum first, only then adding the overheads to fine-tune the mix. However, comparing the results showed that this approach had been totally back-to-front and, from that point on, the techniques became more understandable.

"Once we'd spent the time getting the positioning and sound of the overheads right, we moved onto the snare and kick mics. One thing I hadn't realised, until Mike demonstrated it to me, was how much the phasing of the mics can affect the overall sound. Fine-tuning the EQ and compression on the way into the VS also made a big difference, although the snare, kick and toms seemed to have strongly defined sounds, so it took less work to get them right. Mike stressed the importance of getting the signals into the VS as hot as possible, to make the best of the converters, and comparing with the Blondie track to keep what we were doing in perspective.

"Another thing I learned was to forget dramatic reverbs and delays for my drums, and to work instead with subtle ambience effects on the overheads. Once some close-miked kick and snare had been added to the overheads for definition, a touch of extra ambience made the mix suddenly start to sit together.

"And what of the expensive gear Mike brought with him? Well, we managed to sort things out using just my own kit, so Granny is still around! It's tempting to feel a little smug about that, but I have to acknowledge that 80 percent of my studio has been bought after reading SOS reviews, so my subscription must have paid off! However, if they publish the picture of the PZM mic hung up with garden wire then I'll probably sue them..."

Finishing Touches

At this point we were getting a nice balance of kick, snare and cymbals, including the hi-hat, so we retired to the kitchen for some well-earned lunch! Afterwards we had another listen to the Blondie track to get our bearings and, although we were definitely in the right ball park, we were still a long way away from achieving the large room sound of 'Maria' — hardly surprising really, given that those drums

ms were recorded at a big professional studio! My first instinct was to try to record more of the ambience of the room, so I quickly put up one of Alan's small-diaphragm Shure mics in front of the kit, pointing out into the room to capture more of the room sound. Running this through a Dbx 376 voice channel I'd brought with me, I added some compression and mixed it in with the overheads. Although this made the sound more ambient, it didn't really help increase the size of the perceived space very much, so we quickly abandoned this approach and resolved to simulate a larger room sound artificially.



A bit of rolled-up tissue paper gaffered to the snare drum's head helped to sort out an undesirable resonance.

Alan had a Lexicon MPX100 in his rack and, seeing that I've had good experiences with a similar Lexicon unit in my own setup, I plumbed this unit into the system to see what it could do. We connected the reverb's input to an auxiliary send on the Roland and returned the reverb through the remaining two analogue inputs on the VS880EX — the reverb was fed mostly from the overheads, although a little was also added to snare. With a test recording playing back, we auditioned various Ambience and Room presets first, followed by the Hall, Chamber and Plate patches, but we couldn't find anything to match what we were after, even after experimenting for some time and exercising the Adjust knob. I have to say that I was rather surprised at this — I dare say we could have found a suitable setting if we'd had more time, but editing was too restrictive for us to get results in a hurry. In the end, I resorted to using the internal effects processor in the VS880EX, using the Large Room patch as a starting point. I increased the room size to 22m and pulled the reverb time back to 0.9s to remove any noticeable reverb tail. As a final touch, I increased the density to 60 to smooth out the sound a little, and then adjusted the effects return to a suitable level.

Comparing with our reference Blondie track confirmed that we were much closer to the required sound, although we both felt that a little extra high end was required on the kit overall, so we added a few decibels to the overheads using the Studiomaster's high-frequency EQ, and a few decibels of 14kHz 'air' to the whole kit sound by using Alan's Focusrite Mixmaster patched across the VS880EX's main o

utputs. We also used the Roland's digital EQ to shelf off some low end on the overheads (-5dB at 200Hz) and the snare mic (-5dB at 500Hz), in order to tighten up the low end a bit more.

Even though the overall sound was now pretty well established, the toms were sounding rather distant. Seeing that we still had inputs free on the VS880EX, we put up a spare SM58 covering the two rack toms to see if we could pull their sound forward a little. We plugged the mic straight into one of the VS880EX's mic inputs and, given that the cymbals



were quite low over the toms, I decided to point the mic directly downwards between the two drums to increase the separation. Listening to another test recording showed that the close mic was giving quite a warm tone, where we were after more definition, and also that the toms were ringing in sympathy with kick and snare hits, adding a rather undesirable boomy element to those sounds.

An SM58 was used as a close mic for the toms, and was pointed downwards to minimise cymbal spill.

One of the deficiencies of the VS880 multitrackers is that there is no proper gating or expansion available, which would have been my first choice of processing to remedy the tom resonance — the multitracker's Noise Suppressor insert effect is a threshold-dependent filter which is unsuitable for most drum expansion tasks. In the absence of a gate, I decided to try dealing with the problem using the Roland's digital EQ, rolling off a fair bit of low end and low mid-range: -5dB of low shelf at 400Hz and -5dB of mid-band at 900Hz, with a Q value of one. This reduced the ringing of the toms, but also made the close mic sound pretty thin on its own. Fortunately, the tom sound coming through the overheads was quite warm, so the topky close mic actually helped the overall sound cut through nicely, matching the kick and snare. We didn't have the time to put another close mic on the floor tom, which definitely needed it, but another SM58 fed into the VS880EX's final mic input would almost certainly have done the trick.

An Aphex Anomaly

Before we plugged up the Aphex 109 to process our kick drum mic, Alan noticed that the red overload LEDs on both channels were rapidly flashing on and off, as if high-level LFO signals were passing through them, even though both inputs and outputs were disconnected on the patchbay. I tried adjusting the settings on one of the EQ channels, and noticed that if I reduced the gain on both bands to below unity the light on that channel stopped flashing, which suggested to me that a feedback loop of some type was causing the problem. Plugging into the lower sockets on the patchbay immediately turned the overload LEDs off, which confirmed my suspicions. It turned out that Alan had connected the inputs above the outputs on the patchbay, but hadn't defeated the normalling on those patchbay channels, which meant that the Aphex's inputs were connected to its outputs whenever nothing else was connected, hence the feedback loop.

One Step At A Time

Like many studio activities, the process of sorting out Alan's drum sound involved a series of small steps, with auditioning of the results at each stage to gauge the effectiveness of our actions. Although the differences at each stage were often quite small, the cumulative effect was a radical change in the sound. Switching back and forth between monitoring on headphones (while Alan was playing)

and monitoring on speakers also underlined the importance of doing test recordings when you have no separate control room — you can take an educated guess at EQ and compression settings on enclosed headphones, but it's vital to check everything on speakers as well to avoid any nasty surprises.

Although we'd managed to sync up Alan's two VS recorders (see 'That Sync'ing Feeling' box), I advised Alan not to split his drum recordings across the two machines, because of potential phasing problems arising from the inherently unpredictable nature of the synchronisation. I suggested that he should limit himself to six drum tracks and then use the other machine to record any other band members, being careful to avoid spill from their performances bleeding onto the drum mics. Spill could be kept to a minimum by using the COSM guitar amp simulation on the VS for guitar and bass guide parts, and by placing any acoustic instrumentalists or singers in a bedroom down the corridor from the studio. After capturing the drum parts, I'd be tempted to overdub everything else so that I could reuse the various analogue processors — again, if you make the sounds as good as possible on the way into the VS recorder, then you'll usually get much better results than by trying to fix things in the mix.



If band recording is going to be a frequent activity for Alan, however, I'd suggest that he get himself a slightly more flexible mixer to allow him to set up a decent sound for all the band members together. I'd suggest a four-buss console, something like a Mackie 1604 VLZpro or Behringer Eurorack MX2642A. This would allow Alan to completely avoid the Roland multitracker preamps, the quality of which I've always found slightly questionable, and would give him access to phantom power and three-band EQ on every channel. That would free up his Aphex 109 and Focusrite Platinum units for critical tasks. Both of the desks I've mentioned allow channels to be assigned to the four group busses and master mix buss separately, and these six outputs could be fed straight to the VS880EX for recording the drums. Four extra signals could then be fed to the VS880's analogue inputs from auxes three to six on the Mackie, or from the individual channel direct outputs on the Behringer, leaving at least two mixer auxes for extra foldback, in addition to the two auxes on the master VS880. I'd also suggest that Alan upgrades his outboard reverb processor to something more flexible, perhaps an MPX500 if he likes the Lexicon sound, so that he can more easily tweak the sound to suit the job in hand. **SOS**

A first attempt at increasing the apparent room size involved pointing a Shure small-diaphragm condenser mic out into the room to capture extra ambience, but in the end a more useful result was achieved using the VS880's Large Room effects program with the Reverb Time parameter reduced.

Published in SOS January 2003

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Studio SOS

Tom Fox

Published in SOS March 2003

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People : Studio SOS

Recording in a converted attic, Tom Fox was having serious problems with his acoustics while recording drums, so the SOS team drove over to Yorkshire to sort things out.

Paul White

Tom Fox is a music technology student at Leeds Metropolitan University and he called SOS because he was having trouble recording a decent drum sound in his studio. Given that the recording space is a converted loft, he suspected that some form of acoustic treatment might be needed, and he emailed over a couple of digital photos of the room to give me some idea of what he was up against. After studying the photographs, I guessed a little acoustic treatment would help and, as luck would have it, I'd just been talking to Paul Eastwood of Audio Agency (the UK distributor of Auralex acoustic foam products), who offered a selection of panels and other bits and pieces for Technical Editor Hugh Robjohns and I to take with us.

**Home Studio Acoustics Problems**

We arrived at Tom's home in Yorkshire after a longer than anticipated drive, due to the sudden closure of the M1 following an accident. However, Tom brought out the coffee and a seriously large tin of chocolate biscuits which we started to 'download' immediately, and that soon got us back to normal — whatever that might be exactly! Tom's studio is based around Steinberg's *Cubase VST* running on a 'turbo nutter' 2GHz Pentium IV processor, which he had running smoothly. Monitoring was via a pair of Tannoy Reveal active monitors, while mic preamplification and mixing were courtesy of a Mackie 1604 VLZ mixer. Clearly, technology was not going to be the problem, but the room was certainly less than optimum.

Because of the sloping roof of the loft, side-to-side reflections were minimal, though there were a

couple of boxed-in support beams running the length of the studio on either side at around head height (I should know — I walked into them enough times!) that were fairly close to the speakers and therefore a potential source of unwanted early reflections. More obvious was a strong flutter echo between the flat front and rear walls (the studio was set up along the length of the house) which were completely bare and reflective. The rear wall formed the back of an open stairwell which provided access to the studio, and we decided to start with that.



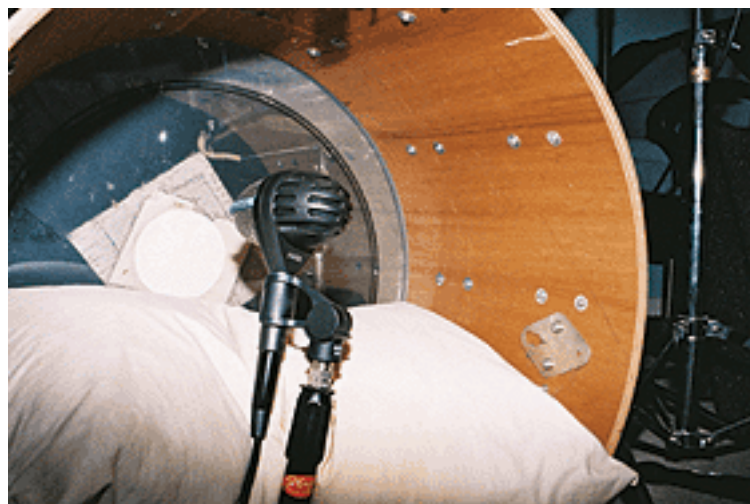
echo could be
mattresses over the
tiles had also been
completely eliminated.
revised by fixing

To deal with the oddly shaped room, Paul installed some Auralex acoustic treatment, fixed in place by spray adhesive.

Tom found a couple of thin single mattresses that we propped/taped in place, and then we balanced a couple of the Auralex foam tiles on top of them to cover more of the wall. This lash-up killed the flutter echo as close to stone-dead as makes no difference, so we decided that, as a permanent job, we'd fix four of the foam tiles to the wall and put the mattresses back where we found them. Two more of the tiles were deployed on the front wall to cut down reflections from behind the monitors and a further two were used on each side of the listening position to cover the side of the supporting beam and an area of the sloping ceiling directly above. We also had a couple of small triangular bass traps, which we used to fill the corners where the supporting beams joined the front wall, but, as the ceiling was sloping, we had to cut the foam to the appropriate angle using a serrated bread knife. The Auralex kit included some rather good spray adhesive, which made fitting the tiles extremely easy, though we discovered it was best to spray both the tiles and the wall, wait around for a minute, then push the tiles into place.

Working On The Drum Sound

Once the foam was in place, the room sounded a lot more controlled and the monitoring environment was much improved. The next challenge was to sort out the drum sound. Tom's main instrument is the guitar — he's only been playing the drums for around eight months — but he was keen to get a good sound so that he could record his own rhythm parts. Tom was looking for a typical rock drum sound, with each drum close-miked and the overheads really just looking after the cymbals. He'd had the kit set up at one side of the room, where the ceiling is very low, and after playing back some recordings that he'd made, it became evident that the sound from the overhead mics was rather coloured, and was also suffering from the reflectivity of the room prior to treatment. The first step was to bring the kit to the middle of the room to give us the maximum possible headroom, though the space we had to play with was still somewhat restricted.



The kick drum had the front head removed and was damped with a pillow, then an AKG D112 was pointed towards the centre of the head.

After moving the drum kit, it became clear that the skins on the toms were relatively slack and so the second step was to check the tuning of the kit, because Tom admitted he had little experience in this area. It turned out that, although the kick and snare sounded pretty good (although the snare was a little dull), the toms were all tuned too low in pitch and the sound was suffering quite badly. While Hugh sorted out the mic cables, I set about tuning the kit and also asked Tom to take the front head off the kick drum, as the hole in the head was too small to allow us to position the kick mic correctly. I don't like leaving the front head off a kick drum for any length of time, as it places an uneven tension on the shell, so I suggested to Tom that he cut a larger hole in the front head and replace it in the near future. The inside of the drum was damped with a pillow in the usual way, so all that was left to do was set up Tom's AKG D112 about half way into the drum on a short banquet stand, pointing directly at the beater — a nice hard plastic one!



All three toms were miked using miniature AKG C418 electret drum mics, which clip directly onto the drum rims, placing the capsule about 2cm above the skin and 5cm in from the rim. They are small enough not to get in the way and the clips are strong enough to hold the mics in place. A flexible bellow in the arm supporting the capsule provides isolation from the mechanical vibrations of the drum. The snare was miked using

a Shure SM57 5cm or so away from the head and angled towards the centre of the drum. For overheads, Tom was using a pair of AKG C1000Ss, which don't have a particularly lively high end. However, they were all we had, so we set them up as best we could in the space available and positioned them to try to give the most even coverage possible of the cymbals, including the hi-hat. That left one AKG C418 spare, which Tom had tried using previously to cover the hi-hat. The revised overhead placement was providing a reasonable amount of hat so we clipped the last C418 to the bottom of the snare drum just in case we could get a better sound by miking both heads together — as it turned out, the SM57 by itself was rather lacking in attack. Normally the lower head mic would be set out of phase with the top mic, but as the Mackie mixer doesn't have a phase switch facility, we decided to record it normally and then use *Cubase's* waveform invert function to do the phase inversion for us on the recorded track. At this stage we fired up *Cubase VST* and made a test recording.



The snare was miked with an SM57 above and an AKG C418 miniature clip-on mic below.

Listening & Adjusting

Listening to the two raw snare tracks together, the combined sound was rather thin and lacking in body, although the snare 'snap' was more evident. Inverting the lower mic track and recombining with the top mic track produced a much fuller sound with both body and attack, and this worked much better. During his previous recording experiments, Tom had been using a gate during recording, but we explained that he'd be safer recording everything straight, then using gate plug-ins if necessary. The reason for suggesting this approach is that a wrongly set gate will ruin an otherwise good take, whereas if you gate after recording you can have as many goes at getting the settings right as you like.

The kick drum sound was very usable as recorded (though we later fine-tuned it using EQ) and the tom sounds were much better for their retuning. Even though the snare sound was much better than when recorded using a single microphone, we felt it could be improved, so we spent a while experimenting. While the lower mic added more high end and snap to the sound, we still felt the overall sound could be brighter and more lively, so I suggested using an enhancer plug-in, as I've often found these to be good rescue tools for lifeless snare drums. Tom loaded up *Isotope Ozone*, which has an enhancement function which we applied from around 1.5kHz upwards and then set the mix balance by ear. I have never used this tool before, so I may not have been making the best use of it, but the end usually justifies the



Because the drums were all close-miked, the overhead mics could be positioned to get the most balanced sound

means in audio. Before long we had a usably from the cymbals. crisp snare sound that would have been difficult to achieve without processing. No further EQ was required, as the enhancer had quite a profound tonal effect on its own.

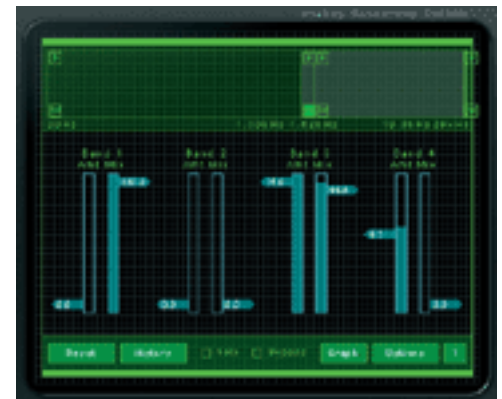
With the whole kit playing, we soloed each track in turn to check for spill and noticed that the floor tom was resonating quite noticeably every time the kick drum was used, so we called up the gate in *Cubase VST's* dynamics section and carefully set it to exclude everything except direct hits on the floor tom. This cleaned up the overall sound quite dramatically, and though the two smaller toms didn't seem to be causing much of a problem, I suggested to Tom that he have a go at gating those too to see if he could clean things up further.

The overhead mics were much cleaner now that the kit was in the centre of the room, but they needed a little EQ presence lift to compensate for the mics' lack of high-end sensitivity — a characteristic of the C1000S. We also rolled off some of the low end, since the overheads were mainly providing the cymbal sound, plus a little ambience, and didn't have to provide the main kit sound as they would in, say, a more jazz-oriented approach. Low end filtering can help maintain clarity and definition when the overheads are added to the close mics, as it helps avoid some of the phase problems that occur when the same drum is picked up by a close mic and by one further away — at the same time, the transients of the drum hits still come through OK. A broad boost centred at around 5kHz helped put some life back into the sound, which just left us with the kick drum to deal with.

As I said earlier, the basic kick sound was very usable, but I wanted to get it sounding tighter, which was easily achieved using a little EQ. By boosting gently at 66Hz, and combining this with a notch at 224Hz, the low end of the sound became tighter and more focused, while boosting at 5.7kHz emphasised the impact click quite nicely. The final settings were arrived at by making adjustments with all the kit tracks switched on — if you work on the kick in isolation, the chances are that it will sound wrong when you have everything else playing.

Final Gloss

The final touch was to find a suitable reverb, so we loaded up Tom's Waves plug-ins and tried *Rezoverb*. By using short decay times of between 0.8 and 1.2 seconds and a bright plate setting, we got a nice sense of space around the snare drum without flooding the sound with reverb. A little of the same reverb could usefully be added to the toms and overheads, but the kick track sounded best left dry. At this point we decided that we had a good enough sound for Tom to customise by experimenting with his plug-ins. However, we did recommend that he get hold of a separate capacitor mic for the hi-hats, as they were slightly lower in level than the crash/ride cymbals, a problem which couldn't be completely cured by



The multi-band enhancer within Isotope *Ozone* was used to get a brighter snare sound.



repositioning the overheads due to space restrictions. In the longer term, the C1000S could also be replaced by mics with a better top end for use as overheads, as there are now several low-cost capacitor microphone models to choose from that would do this job well.

This screenshot shows the EQ settings which were arrived at for processing the kick drum track.

To round off the day and to help compensate for the long drive up, we adjourned to a nearby Indian restaurant and talked shop until the rush-hour traffic had subsided. There, Tom told us that he'd found the visit instructive and he was also very pleased with the improvement that the acoustic treatment had made to his room. **SOS**

Thanks to Audio Agency (www.audioagency.co.uk) for supplying the acoustic treatment used for this article.

Tom's Comments On The Session

Paul and Hugh arrived with a rather large box of Auralex acoustic foam and several cans of liquid glue. Immediately they spotted the annoying flutter echo I was getting between the back and front walls of the room. This was the main problem, because it made accurate monitoring an absolute nightmare! Without a doubt the room is now so much better acoustically and that flutter echo has been banished.

Moving the drums into the centre of the room made them sound more acoustically dead, with much less coloration on the overheads. Plus, I can get some height between the mics and cymbals now. I also heard very noticeable improvements as soon as we had moved and retuned the drum kit — this is how a drum kit is meant to sound!

The visit really did open my eyes to the possibilities and results that are achievable by anyone working in a small project studio like mine. With time and a little patience, and without spending a load of cash, anyone can produce good-quality mixes at a fraction of the cost of paying for time in a commercial studio. A big thanks to Paul and Hugh for taking the time out to come and help me — I really enjoyed it!

Published in SOS March 2003

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SOUND ON SOUND

The World's Best Music Recording Magazine

Studio SOS

Dave Stevens' home studio

Published in SOS May 2003



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People : Studio SOS

This month the SOS team visit Dave Stevens' home studio to sort out a chesty vocal sound, investigate a mystery digital buzz, and hand out some mastering tips.

Paul White

Dave Stevens has a home studio in Gloucestershire where, rather than simply set up a desk for the gear like many of us would, he's divided one of the bedrooms in his bungalow to provide a control room and drum/vocal booth. The soundproofing between the rooms is very effective and the application of Primacoustic foam panels and bass traps in both areas has avoided the boxiness that can occur in small studios. The live room contains a Roland TD8KV drum kit, which David has learned to play remarkably proficiently in a relatively short space of time, plus he has a decent collection of guitars and basses which he tends to record via a Line 6 Pod Pro and Bass Pod Pro.

The Control room is a compact 11.5 x 7 feet, with a window at the back of the room and the speakers set up on the long wall dividing the studio and live room. The Yamaha NS10M studio monitors are on stands either side of a computer table which supports a pair of 19-inch CRT monitors and a Houston moving-fader control surface. All the outboard gear is in a tall homemade rack in the middle of the right-hand wall of the studio. A double-glazed control-room window provides a clear view of the live room where David has a Neumann U87 microphone set up for vocal recording (with a homemade support constructed from a piece of nylon tube!). Recording is to a PC, which David built, running Steinberg *Cubase SX* and using Yamaha DS2416 and SW1000XG cards. The analogue and digital



Paul White listened to Dave's recorded vocal tracks and attempted to remedy the congested sound using EQ plug-ins.

However, this strategy proved unsuccessful, so they decided to tackle the problems at source by rerecording the vocals from scratch.

inputs and outputs from his soundcards are wired to a patchbay and digital router respectively, allowing signals to be processed externally and returned to the *Cuba*



se environment. For example, Dave has *Cubase SX* set up to make use of his Lexicon MPX550 hardware reverb, which then feeds back into *Cubase's* virtual mixer. A Spirit analogue mixer provides input mixing and routing for the various sound modules to *Cubase*, and acts as a monitoring controller.

Improving The Vocal Sound

Dave contacted us because, although he felt he was recording everything pretty well, his mixes didn't have the sparkle he felt they should have. He was also having problems getting the vocal sound he wanted. Furthermore, it transpired that he was suffering from a low-level buzz which he believed came from his SW1000XG soundcard (and which disappeared temporarily when scrolling graphics on the screen!), but was at a loss as to how to cure it -- more of that later.

When we arrived, Dave produced a vast quantity of Marks & Spencer's exec

utive-grade chocolate biscuits and countless mugs of coffee, so we felt we had to give it our best shot! After locking Dylan the dog out of the studio, we set about listening to a song David was currently working on and which featured his own vocals. Everything was recorded cleanly, but the vocals had a very chesty, congested sound, so in the first instance I tried to find a parametric EQ setting (using a Waves plug-in) to fix this. However, even though notching out the worst low mid-range peaks helped, it was clear that this wasn't a proper solution, so we asked David to explain how he'd recorded the vocals.

His system was to use the U87 at around six inches, with a pop shield fitted between him and the mic. The mic fed through a Focusrite Platinum Voicemaster Pro, where he applied what he felt was a gentle amount of compression, a little mid-range cut to remove some of the chestiness, and the simulated tube section to warm the sound up a little. So far it all made sense, but then David said that he also passed the signal through his Antares ATR1 (the hardware version of the *Auto-Tune* plug-in) to tune it before recording it (though he monitored the non-corrected



signal, as it's almost impossible to sing when monitoring the corrected output). He'd also



As a final touch for the vocal sound, a little reverb was applied using the Warm Plate patch within Dave's Lexicon MPX550.

everything, removed the ATR1 from the loop, and then asked David to sing while we listened to him in the control room. Dave had left the voicemaster set up exactly as he had it when recording the original track, and the first thing I noticed was that the level settings David had on the Voicemaster Pro were a bit on the high side, so I backed them off to leave us with a reasonable amount of headroom.

With no compression or tube simulation, the sound was much less congested, but it still sounded too chesty to me, so we asked David to move back to around 18 inches from the mic. This brought about a further improvement, but not a complete cure, so the next step was to drop the mic height so that David was, in effect, singing towards the top edge of the pop shield. By using this mic position and working at between 12 and 18 inches from the mic, we achieved what we felt was closer to David's natural vocal sound, and it was certainly a lot nicer to listen to than his 'before' version!

Though I often add a small amount of compression while recording, I felt that in this instance David's levels were well enough controlled that we could allow ourselves the luxury of recording completely flat, leaving ourselves the option to use plug-ins to d

o whatever was necessary after recording. I also suggested that *Auto-Tune* should be applied as a plug-in after recording, as that gives the engineer a chance to optimise the settings for the song, or even to divide the song into sections and process different sections separately where appropriate. If it is used when recording, inappropriate settings could destroy an otherwise good take.

We also avoided using the tube simulation, because, although this can be most effective for thickening up thin voices, it can have a congesting effect on a vocalist who already has a strong lower mid-range -- which Dave did. Using this setup, we recorded a new vocal part and then tried to sit it in the mix.

Because Dave was singing only about six inches away from his Neumann U87, the mic's proximity effect was contributing to the chesty sound. It also transpired that Dave had been doing a lot of processing to the signal before it was recorded, so all of this processing was bypassed so that more options were left open for the mixdown stage. In the end, some compression was all that was required, set up on Waves' *Renaissance Compressor* as shown in the screenshot.

o used the *Cubase SX* host-powered reverb plug-in (which didn't have the right quality for his voice) rather than

his Lexicon. Finally, Dave was also using *Cubase's* True Tape emulation mode with 12dB of overdrive, to try to create a more analogue quality.

The only real way forward was to get David to sing the track again with us at the controls and see what we could do at the tracking stage. To this end, I bypassed

As it turned out, no EQ was thought necessary, though compression was added using the Waves *Renaissance Compressor* with a ratio of around 6:1 and the threshold set to give about 8dB of gain reduction on the loudest notes. A fairly fast attack time was combined with a medium/fast release time -- the exact settings can be seen in the screen shot. For the finishing touch, we added some reverb from the Lexicon MPX550's Warm Plate preset, tweaked for more pre-delay (around 80-90ms worked fine) and the decay time was set to 1.6 seconds.

Hugh was concerned that David had also been recording with the Cubase True Tape facility active, which essentially provides soft saturation on higher-level sounds (he had the threshold set at 12dB from maximum). It was just possible that this degree of saturation, combined with the valve emulation on the Focusrite preamp, might have contributed to the thickening of the vocal sound, so we ran two vocal recording tests, one with and one without True Tape. As it turned out, there was no material change in timbre so True Tape was exonerated!

Dave's Soundcard Problems

Background buzzes and ticking sounds that are affected by mouse movements and drive accesses can be caused by a variety of mechanisms, so it pays to be systematic. I started by checking that Dave didn't have an unusual graphics card, since a few types (Nvidia is one manufacturer that springs to mind) have been known to induce soundcard interference, as have some Adaptec SCSI cards. In these cases it's worth trying to move your soundcard to a slot as far away from the offending item as possible. However, he no longer used SCSI, and his Matrox dual-head graphics card (identical to mine) has an extremely good track record for musicians. Dave had also taken the often wise precaution of placing his Yamaha DS2416 and SW1000XG soundcard pair well away from the AGP slot anyway, leaving two empty slots in between, so I didn't need to try this either. I next checked that all cards had been well seated in their slots and the backplate screws tightened down firmly to give them a solid ground connection, but again everything was in order.

Interconnected Yamaha DS2416 cards employ a ferrite RF filter on their short interconnecting ribbon cable to minimise interference pickup from other internal components, but the internal cable used for connecting a DS2416 to the SW1000XG

Roland TD8KV Drum Kit Tips

A final tip, and nothing to do with vocals, concerned the Roland TD8KV electronic kit Dave was using. The sounds that David had chosen were probably fine in isolation, but in the mix they lacked definition, so it pays to choose your sounds in context rather than just picking what you feel is a nice sounding kit. In particular, the kick drum was not very well defined, so I suggested trying the one from the heavy rock kit, which has more weight and definition. My other tip was to buy and use a real hi-hat -- the hi-hats are the one thing that lets down all the electronic drum kits I've played. The physical noise from the mesh heads used on these drums is low enough that it shouldn't interfere too much with miking a hi-hat.

doesn't, so I'd taken the precaution of bringing one along to try. However, it made no difference to the audible interference after clamping it in place, and neither did an impromptu screen placed round the soundcards -- kitchen foil inside a polythene bag to prevent it shorting something out, with the aluminium foil temporarily connected by an earth wire to the PC metal work. Even temporarily disconnecting all Dave's USB peripherals (which have been known to cause the occasional hum) made no difference.

We hit a glitch when we rebooted Dave's PC with his Houston controller still switched on -- the computer suddenly became unable to see his RAID (Redundant Array of Independent Disks) setup and his system hung. However, he'd already suffered from the same problem on his IMSI 845E motherboard a couple of times previously, and passed on details of how to cure it just in case anyone else ever experiences anything similar. You turn off Houston, reboot, and then enter the RAID BIOS during startup. Then you delete the current drives and set up exactly the same RAID array once more. It seems that the order in which you switch on peripherals is crucial for this particular system.

We continued checking at the PC end for possible reasons for the annoying amount of background noise, including the SW1000XG's inputs being accidentally left at mic sensitivity and confirming that the external cables had been correctly wired with good shielding, but eventually we were all satisfied that there was nothing else to try at the source end, and we could only assume that this was a more general ground loop problem.

Martin Walker

Buzz Busting

Having obtained a better vocal sound, we set about finding a strategy to more effectively master David's mixes using the available plug-ins. However, in the course of our listening tests a rather annoying digital-sounding buzz which emanated from his DS2

416 soundcard's analogue outputs had to be addressed. This initially appeared to be a software driver problem, particularly since the noise would disappear momentarily when the computer was doing other things (such as updating the screen), although Dave had already updated all his video and soundcard drivers.

As our colleague Martin Walker has experience with such problems, and because he only lives around 20 minutes from David's house, we asked him to come round and check over the system. While we

waited for Martin to arrive, we started checking to see just which outputs suffered the noise and discovered that the digital outputs from the card seemed clean, so we started to think the problem might involve ground loops, rather than software problems. To test this idea we connected one of the sound card's analogue outputs directly to the monitoring input on David's TD-8KV drum kit -- the handiest source of headphone monitoring which was also isolated from the rest of the rack's earthing. Sure enough, this turned out to be clean too, so there was obviously a ground loop issue here.



Before getting down to remastering some of Dave's tracks, an annoying digital-sounding buzz coming from the monitor speakers first had to be eliminated. After much detective work, Paul decided that earth loops in the monitoring chain were to blame -- the buzz disappeared when listening to the output of the soundcard directly through a headphone amp isolated from the main rack's earthing.

Making up balanced cables to feed the power amplifier from the Spirit mixer's monitor outputs improved the situation noticeably, but the buzz was still there and, perversely, it disappeared if the audio was routed via the outputs of the SW1000XG, even though the grounding regime was essentially identical. We tried all the standard ground loop fixes, including unplugging all the mixer connections and then putting them back one at a time, but we couldn't find a definitive cause. I even tried the old 'resistor in series and lift the screen' trick on the soundcard outputs, which ended up being like an episode of Junkyard Challenge because nobody had brought any resistors. I toyed with the idea of fixing two electrodes to a known volume of coal, but eventually settled on using the resistance of a lead pencil -- all of which went to prove that the problem was still there!

When Martin arrived he checked all the computer's internal connections, tried installing a ferrite clamp and adding additional screening between the cards (see the 'Dave's Soundcard Problems' box for details), but David had already done just about ev

erything that Martin could recommend trying. Then disaster struck when David patched everything back in after Martin's exploratory efforts, because the whole system went into a high-frequency howlround. Before we could turn down the volume, jets of smoke shot out of both of David's NS10 tweeters. One had died altogether and the other, while still appearing to be work OK, was clearly not a happy tweeter! Fortunately, Peter Peck at Yamaha Kemble was able to save the day, because, when we phoned him to enquire about replacements, he graciously arranged for a pair to be sent to us free of charge. They arrived just two days later, so a big thanks to Yamaha for bailing us out!



One-legged Mastering

Having sorted out the levels and checked the wiring, we continued with the one injured, but still working, tweeter to find a mastering solution. As mastering with one blown and one lightly toasted tweeter isn't to be recommended, we also made CDs of our efforts and checked them on David's hi-fi system in the lounge. What we came up with was predictable enough, but, although David had been using a similar collection of plug-ins to do the job, he hadn't

appreciated the benefits of using low-threshold, low-ratio compression when mastering to reduce the overall dynamic range in a subtle way before applying a peak limiter. He'd treated his mixes in a similar way to his individual tracks, and had also been a bit heavy-handed with the limiter -- while his mixes didn't actually sound that bad, they didn't have the transparency and sparkle he was looking for.

I assembled a Waves *C1* full-band compressor, a Waves *Renaissance EQ*, and a Waves *L1 Ultramaximizer* limiter, with dithering set to 16 bits. We also tried the Steinberg *Spectralizer* harmonic enhancer and found that it could produce useful results in some mastering situations -- to



get the benefit of the dither and limiting, however, we patched it prior to the *L1*. The settings we tried included compression ratios of 1.1:1 and 1.2:1 with the threshold pulled right down to give a level reduction of around 6dB on the loudest sections. The attack was fairly fast, with the shortest release time we could set without invoking pumping -- you can see the settings we ended up with for all the processes in the screenshots.

The EQ was set up to give a very gentle 'smile' curve, by adding a hint of broad boost to the frequency extremes, while the middle was dropped by a decibel or two

using a very wide, gentle setting. This is a common ploy to add clarity and sparkle to a mix. Where the *Spectralizer* was used, the amount of harmonics added was carefully monitored to ensure that only a slight brightening effect was achieved -- it is so easy to go too far with these things. To finish off, the limiter output target level was set to just below maximum (around half a decibel is a common setting) and the input gain was adjusted so that the amount of limiting on the peaks was only between three and five decibels. As we had already reduced the dynamic range with the compressor, most of the time the *L1* gain-reduction indicators showed little or no gain reduction taking place -- whereas when David had used it, it was showing gain reduction almost all the time.

The result of this simple plug-in setup was a general levelling of the track and an increase in density, but with more transparency and detail in the mix. The final track was also hugely louder than the original stereo mix, even though that was peaking close to full scale on several occasions. But,

importantly, the new mix didn't sound squashed, even though it was louder. The tip here is that, when mastering, don't try to get all your level gain by limiting, as that just squashes the peaks and does nothing to improve the density of lower-level sections. By using low-ratio, low-threshold compression, the whole mix sounds more homogenised and, as a bonus, it leaves the limiter with less work to do. This technique also shows that you don't have to have a multi-band compressor for mastering and, indeed, many mastering engineers seem to prefer using a full-band compressor anyway. Multi-band compression does help get the maximum possible level, but it can be at the expense of tonality or balance if used to excess.

At the end of our day with Dave he asked for any other advice or comments, and Hugh came up with a couple of suggestions to improve the acoustics in the control room. The first was to replace the large CRT monitors with LCD screens -- although this would be a fairly expensive option because Dave was so used to his 19-inch monitors that only 17-inch flat screens would probably do! The reason for changing the monitors was primarily because of their intrusion into the monitoring area between the NS10 speakers, causing reflections and scattering of sound which had a harmful effect on the stereo image. This situation would only be exacerbated when Dave installed his keenly awaited Mackie HR824 full-range monitors alongside the nearfield Yamahas. Using flat-screen monitors would also benefit his guitar recording by reducing the amount of electromagnetic radiation likely to be picked up by the guitar.

A similar acoustical problem was caused by the placement of the equipment rack, which was very close to the right-hand speaker and caused more reflections and sound scattering. By placing the rack closer to the rear corner, this situation could be remedied very easily, and Dave actually achieved this shortly after we had left. **SOS**



Locating the source of the mystery digital buzz turned out to be quite a palaver, so an emergency call was made to SOS PC specialist Martin Walker. A variety of checks were made on Dave's system, confirming that it was well set up for music use -- not only were his applications and drivers up to date, but his audio cards were also positioned well away from his graphics card (top) and were well seated in their slots.

Published in SOS May 2003

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Studio SOS: Nick Redman

Published in SOS June 2003

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People : Studio SOS

The SOS team battles through snowdrifts to help Nick Redman and Mike Sinnott with two different vocal sounds.



Paul White

This month's tale of intrepid studio rescue takes us back to the end of January, and sees Technical Editor Hugh Robjohns and myself battling blizzards and navigating the M25 in first gear to reach Nick Redman's bedroom studio near Croydon. Nick had called us asking for help with his vocal mixing, because, although he had good equipment and some decent vocalists, he was never happy with the vocal sound or the way it sat in the mix. Along with his partner Mike Sinnott (who looks after student welfare at the Alchemea recording school), Nick composes and records mainly dance music, but to gain experience and to help pay for his well-stocked studio, he also works with a number of paying clients. This means he has to deal with various vocalists and what he needed was a strategy to help him fine-tune each one to its best advantage.

Cubic Quandary

Nick's studio was a very unfortunate shape, being an almost perfect cube around nine feet along each side. This was bad from an acoustic point of view, as the room modes all stacked up at the same few

frequencies where ideally you need them to be as evenly distributed as possible. Only the bass trapping effect of the window saved it from low-frequency chaos, while a few strategically placed foam tiles calmed the mid-range and high-frequency reflections as well as providing a reasonably dead corner where the vocals were generally recorded. The studio's Mackie HR824 monitors were set with the bass switches in their middle positions and a pair of NS10s were available for alternative monitoring. A Soundtracs Project 8 mixer looked after mixing the outputs from *Cubase SX* running on a 2GHz Pentium 4 machine (Windows XP) with extensive silencing mods — Nick is an IT engineer, so this part of the system presented him with few problems. There was also a whole rack of well-chosen outboard including an Eventide Orville and an Avalon VT737 tube voice channel. The PC was also fitted with a Universal Audio UAD1 card and had a full bundle of Waves plug-ins.

The first step was to ask Nick how he went about recording the vocals. He explained that the singer stood with his or her back to the treated corner and sang into either a Neumann U87 or an AKG C414ULS through a pop shield. The house is right next to a busy commuter rail line, so Nick usually waited until a train had passed, then tried to get a take down before the next one turned up! He also used high-pass filtering to keep low-frequency rumble to a minimum, although he also used good-quality shockmounts with the mics.

Standing with their backs to the corner in the confined space of the room meant that the singers couldn't back off very far when singing loud passages, but otherwise it seemed to work OK. The mic was normally fed to *Cubase* via the Avalon unit, which was used to apply the bass roll-off, as well as a little gentle compression just to keep the peaks under control. When mixing, Nick fed the vocal track back through the Avalon so that he could use its EQ and compressor, adding vocal reverb from the Orville.



Although the equalisation from the Avalon VT737SP was great for smooth-sounding tonal shaping, it wasn't sufficiently surgical to tame troublesome narrow-band resonances.

My only suggestion about the recording environment was to fit a curtain pole or shower curtain rail to the ceiling and then hang a duvet from it to divide the vocal corner from the rest of the room. This would further cut down any spill from the computer drives and fans and also reduce coloration caused by room reflections. Nick felt that this was a good idea that wouldn't cost much to try out, but in deference to our art department, I declined to be photographed holding up a duvet yet again!

Sweetening A Female Vocal Track

As Nick and Mike work with a number of different clients, I suggested that we check out two mixes that were causing particular problems and then see what we could do to improve them. The first track featured a female singer with a very attractive voice, but as soon as she hit the chorus the dynamic of

the vocal rose dramatically causing very high peaks and a shift in timbre from very sweet to very forceful. Although this was the intention for the mood of the song, the shift in tone introduced some harsher frequencies in the recording which Hugh and I felt created an over-strident timbre. Although there were plans to re-record this vocal part in the near future, I didn't feel that the timbre would be much different, and a comparison recording of the same vocal part made at the Alchemea studio showed the sound to be almost identical, so there was clearly nothing much wrong with the basic recording technique.

The dry recording was clean enough, but the Eventide Orville plate reverb setting Nick and Mike had chosen sounded too splashy to us. Also, while it was obvious at higher settings, it almost disappeared when it was turned down in level, leaving the vocal sounding dry and thin. As you can only really work on one thing at a time, we switched off the reverb and concentrated on getting the tonality of the voice right. After trying to tame the offending frequencies using the Avalon's EQ, it became apparent to me that it was the wrong tool for the job. While it is a lovely-sounding EQ that can be used to add warmth, air and polish to a sound, I felt it was too polite for the task in hand, where we needed to apply some pretty aggressive notching to those frequencies causing shrillness. Furthermore, because of the energy being put in by the vocalist during the choruses, the vocal character was very different between the verse and chorus, so I suggested to Nick that he shift the verses and choruses to separate sequencer tracks so that we could more easily treat them differently without having to mess around with plug-in automation.

As the chorus was the most difficult section, I asked Nick to load up his Waves Q4 equaliser so that we could address any problematic frequency peaks, then we looped around the offending section while making adjustments. I used the usual technique of applying maximum boost while sweeping through the frequency range searching for the problem frequencies, and it soon became apparent that there was a very aggressive peak at around 3.8kHz, with another peak an octave below that. Deploying two notch filters with pretty large amounts of cut tamed these quite effectively — the exact settings can be seen in the screenshot. To avoid dulling the sound, and to add a little 'air', we used some gentle boost at 16kHz, and we also tried to warm up the lower mid-range by adding around 1.5dB of boost at 250Hz with a very low Q value. The overall consensus was that this improved the chorus vocal dramatically, making it far less strident, adding warmth and body, and at the same time keeping the sense of air and detail. The verse vocal wasn't nearly so strident, so we copied the EQ setting to another Q4 plug-in and then reduced the depth of the cut notches until a musical sound was achieved without the verse and chorus sounding



The compression and EQ settings used for the female chorus vocals. The two narrow-band notches on the Waves Q4 plug-in were used to tame harshness, and the compressor was showing about 6-7dB on the loudest peaks.



For the female verse vocals, similar EQ settings were used, but with reduced gain settings. The compression was also backed off a little, giving only 3-4dB of gain reduction on signal peaks.

mismatched in any way.

Reverb & Compression

That left reverb and compression to sort out. The vocal had been compressed to some extent while recording, so we used the modelled 1176 compressor plug-in that comes with the UAD1 card, setting a fast attack and a ratio of 8:1. We adjusted the threshold so that the compressor came in on the louder parts of the chorus (showing a gain reduction of 6-7dB) and set the release control to around 25 percent. This helped even out the levels, and also gave the sound a little of that 1176 character. A similar setting worked with the verse. However, given that there were fewer peaks, the amount of gain reduction could be reduced to 3-4dB.

Now it was Hugh's turn to try to find a better reverb sound. We wanted something with strong early reflections after a longish pre-delay to provide some character and support to the voice, but with a gentle and low-level reverb tail that wouldn't get in the way. The original plate program offered very few parameters for tweaking, and none of the quick tweaks made to alternative Orville programs provided the result we wanted. It may have been possible to find something suitable given enough time and familiarity with the Orville, but as we had neither in abundance I suggested that we try the UAD1's *Realverb Pro* plug-in, something Nick and Mike had shied away from as they'd always assumed outboard reverb would be better.

Nick set this up to return to two channels on the desk, giving us quick access to the return levels, then I had a go at creating something suitable — as I hadn't used a *Realverb Pro* before, I was keen to see what it could do. Manipulating the various elements of the reverb, I managed to create a reverb setting with a strong early reflection pattern and a decent amount of pre-delay to fatten and thicken the sound without making it too wet. On top of that, I added a reverb tail around three seconds long, but brought its level down with respect to the early reflections so that the overall result wasn't too washy.

With a little adjustment to the EQ and 'wall materials', we soon came up with a nice treatment that added real weight to the voice and gave it a sense of existing in a real space without the splashy



Paul experimented with reverb using the *Realverb Pro* plug-in on Nick's Universal Audio UAD1 card — the final settings can be seen here.

wetness we'd suffered from originally. Bringing up the rest of the backing confirmed that we'd improved the situation significantly, with the vocals sitting much more comfortably in the mix. Hitting the Waves EQ bypass button brought back a stark reminder of just how strident the original vocal track had sounded in the chorus and everyone agreed that the sound we'd finally arrived at was a lot sweeter and much closer to what Nick and Mike had originally been aiming for.

Fixing Boxiness In A Male Vocal Track

The second track we heard featured a male vocal which was well pitched and carried plenty of expression, but again it didn't sit well in the track, even though the overall level wasn't bad. My immediate impression was that the voice sounded boxy and congested. Also, as with the previous track, the chosen reverb treatment didn't get it to sit naturally with the backing. The boxiness was tamed by using Nick's Waves Q10 equaliser to apply gentle notches at around 500Hz and 2kHz, and again this was balanced by a little low-end lift (this time right down at 63Hz, but with a very wide bandwidth) and the obligatory 'air' EQ at 16kHz. The difference was dramatic, and right away the vocal felt better in the mix, even before we started on the reverb treatment.

Compression was applied using the UAD1's 1176 emulation, again in a fairly gentle fashion with no more than 6-7dB of gain reduction on the loudest notes — this was quite enough to keep the vocal level stable in the track, and again you can see the exact EQ and compression settings in the screen shot. For reverb, we ended up settling on the *Realverb Pro* again, with just a few minor tweaks, the main point being to add strong early reflections, once again to give the vocal weight and space.


The Moral Of The Story

The main thing that came to light from this challenge was that not all high-end EQs are suitable for all jobs, and in this case the Q4 and Q10 were far better suited to surgical notching than the mellifluous Avalon. It also pays to keep in mind that you can get away with quite drastic EQ cutting without the processing itself becoming obvious. In most instances, though, boosting has to be gentle and done with high-bandwidth settings to avoid sounding phasey or nasal. Even when a voice is perfectly recorded, there may be physiological resonances that are best attenuated, so although EQ shouldn't be considered mandatory, you shouldn't be afraid to use it when it is needed.

Additionally, we demonstrated that you can't be over-casual with the choice of reverb and in many



Compression and equalisation were used to deal with a boxy male vocal track, and then reverb was used to add weight and to get the vocal to sit well in the mix.

cases a little basic editing is necessary. Part of the secret is knowing what you want the reverb to achieve, and in this case we needed the early reflections to add weight to the voice as well as to provide a convincing sense of location. The reverb tail is also important, but it doesn't have to dominate the sound, and in modern styles where the vocal needs to have a less obviously reverberant sound, keeping the reverb tail down below the early reflections can help a lot. 

Published in SOS June 2003

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Studio SOS

Allan Murrell

Published in SOS July 2003

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People : Studio SOS

This month the intrepid SOS team travel to Wigan to address Allan Murrell's recording, monitoring and mixing problems.

Paul White

Allan Murrell has a very enviable studio setup made all the more attractive because it is set up in a spacious farmhouse near Wigan where he has a separate control room, a spacious live room and a smaller adjoining isolation room with a window onto the live room that can be used for recording vocals or drums. These are all repurposed from an entrance porch, dining room and living room — apparently to the dismay of his long-suffering girlfriend! Rather than go the computer route, he has based his system around a Mackie HDR24/96 24-track recorder, a Mackie D8b mixing console (with numerous plug-ins) and a pair of Mackie HR824 active monitors, backed up by a pair of Genelec 1029As. Just in case you think he's bought shares in Mackie, he also has a rack of outboard processors that includes a Lexicon MPX500 reverb, a TC Electronic M*One XL reverb and a TC-Helicon Voice One vocal processor, as well as some nice analogue boxes, including a TL Audio Ivory 2 valve compressor.

He also has a good selection of mics, including the budget Superlux drum set, an AKG D112 kick-drum mic, some Rode NT3s and (his current favourite) a Rode NTK tube mic. Though Allan set up the studio for his own use, it soon became clear that there was a demand from bands and artists in his area, so the studio is becoming increasingly commercial. To enable bands to get good results very quickly, the live room is set up with a Roland V-Drum kit (with real cymbals and often a real snare drum), two Line 6 Pods and one Line 6 Bass Pod, the latter three set up on stands with headphones for the players. There is also an acoustic drum kit for those that require or prefer it.

Allan called us in because he'd had some problems with his mixes not sounding as good on systems outside his studio, and he was also concerned that his small live room made vocals sound slightly coloured. Additionally, he wanted us to check over some mixes he was working on, as he felt he wasn't getting the best possible sound from all the tracks and wanted to see if he could get cleaner-

sounding mixes. After a surprisingly uneventful trip up the M6, Technical Editor Hugh Robjohns and I found ourselves sitting in Allan's control room drinking coffee and negotiating a plate of chocolate cake — Allan had clearly read previous reports and was aware of the requirements regarding esurience displacement!

Control Room Troubleshooting

We decided to start with the large control room, which was rather unfortunately proportioned, being almost 14 feet square and a little less than half that in height. Having dimensions that are close to being equal to or multiples of each other tends to exaggerate room mode problems, though the two large windows and one door helped to act as bass traps here. Before suggesting anything, we listened to a selection of Allan's mixes and some commercial CDs through the system before deciding on a course of action. The room tended to produce a slightly muddy sound with less-than-ideal stereo imaging, and one or two bass notes really set off the room's standing waves.

Part of the problem was that the Mackie monitors were standing on outrigger shelves on the IKEA computer table that formed the control centre for the studio. These were not particularly rigid shelves, and even though the speakers were standing on dense foam blocks, both Hugh and I thought the bass end could be tightened up by mounting them on proper stands. We also felt that reflections from the wall behind the monitors should be killed if at all possible, something we tested by first fixing a duvet to the wall, then adding some profiled foam sheeting when we confirmed that it did indeed help tighten up the sound, particularly the stereo imaging. As Allan had plenty of spare foam sheet, we also fixed a couple of panels to the ceiling above and slightly in front of the mixing position (using spray adhesive), to damp down reflections from the low ceiling.

Comparing the Mackie monitors with the pair of Genelec 1029As Allan had for reference, we came to the conclusion that the Genelecs were sounding rather boxy, with too much bass, while the Mackie monitors were slightly bass light in their current position well away from the back wall. Setting the rear-panel switches on the Mackies to normal for the high and low ends (rather than the middle setting for the bass switch) and leaving the speaker environment switch set to 'full space' proved to work rather well. Hugh needed to set about the 1029As with a screwdriver to move the heavily recessed DIP switches to setting three so as to introduce around 2dB of low cut starting at around 500Hz. We also moved the Mackie monitors inwards towards the support side of their shelves, to try to improve their stability, moved them forward and angled them inwards, aiming at the back of the mixing chair.

Because Allan had read one of our previous Studio SOS pieces where there had been problems with sound from Genelec 1029As reflecting off the computer monitors, he had installed more foam



Reflective surfaces in the control room were causing Allan monitoring problems. Following the obligatory experimentation with duvets hung in various locations, it was decided that absorption was required not only behind the monitors, but also on the ceiling above the monitoring position.

between his monitors and speakers. However, in this situation there was actually little risk of reflections, since the speakers were slightly in front of the monitor screens with this particular setup.

Once all these adjustments were complete, the sound was noticeably tighter, and the two sets of monitors were in closer agreement as to how the mixes should sound. A duvet was hung over the door in the side wall to damp refl

ections, while the curtains were drawn on the window opposite to cut down on reflections from the glass (as well as sunlight causing glare on the computer monitors). We also suggested moving the entire equipment rack back a little closer to the rear wall, but as it needed to be completely stripped down before it could be moved, we left that for Allan to try at his leisure...

Live Room Acoustic Problems

The small live room presented a different problem, as our speech immediately sounded coloured when we walked in there, even though almost all the walls and windows were covered in more of Allan's profiled foam. I guessed that part of the problem was down to the foam being fairly thin and very lightweight, with a fairly open cell structure, which meant it was probably only absorbing effectively above 500Hz or so, mopping up all the high end but leaving the lower mid-range and bass frequencies to reflect freely. Actually the large window area probably helped with the deep bass, as most of this would pass straight through, but frequencies in the 150-350Hz region were definitely dominating the room. Allan was also recording with the mic set up very close to one wall, so we suggested removing the foam from some of the window area at one end of the room and using it to double up the foam at the other end to form a well-damped corner where the singer could stand (back to the corner). We fixed up the foam in a temporary fashion, moved the mic stand, and straight away the sound was more open, with a better balance of mid-range and high frequencies.

We also felt that replacing the foam on the wall opposite the window with a thicker, more dense type



Although most of the surfaces around the vocal mic were covered with open-cell foam, the relatively thin foam used only affected the sound above 500Hz, making for a rather coloured vocal sound. Doubling up the foam thickness in this case helped to absorb more low frequencies, providing a more balanced sound.



(three to four inches thick) would control the low mid-range even better. However, as nearly half this wall was given over to the window looking out onto the live room, this wouldn't kill any flutter echo between the two windows.

As predicted, there was some noticeable flutter echo at the opposite end of the room to the vocal corner, where we'd removed the foam, which might be a problem when recording drums, but the vocal end was fine. As Allan wanted to get better sound isolation between the big live room and the small one, we suggested that, as the house walls were very thick, he could remove the single-glazed window dividing the two rooms and replace this with a double-glazed one, set at an angle to the window opposite. This would kill the worst of the flutter echo and significantly improve the isolation. As Allan had a double-glazing engineer as a studio client, this seemed to be a very promising avenue!

We also noticed that a lot of fine dust was coming off the foam, and this turned out to be the polyurethane breaking down due to the action of sunlight. Where such foam is to be used in direct natural light, it is best to cover it with fabric or, where it is to be used in a window opening, to fix it to hardboard or MDF with the board facing the window. Painting the board black provides a professional and neat finish for anyone looking at the window from the outside.

Fixing The Mix

Allan had been working with a rather good band that clearly had Pink Floyd/Roger Waters influences and, though his mix didn't sound bad, it didn't have the punch and clarity that he wanted. My first experiment was to try the Acuma Labs *Final Mix* plug-in on the D8b to attempt some mastering-type processing, which involved low-ratio, low-threshold three-band compression and a touch of the inevitable 'air' EQ (gentle wide boost at 16kHz in this case).

I reset the processor's crossover frequencies to 150Hz and 5kHz so as to leave the mid-band intact and used the gain settings in the three compressor bands to fine-tune the tonal balance. This opened up the mix quite noticeably, but there were some track EQ and balance issues we felt we could improve on, so all three of us went through the mix a channel at a time and checked the quality of the basic sounds while looking at the processing Allan had used.

Allan felt he had got into a bit of a rut with his EQ settings and was applying some 'by habit' curves in situations where they might not be the best thing to use. Furthermore, he had used some compression and EQ while recording and, while this can be perfectly valid in some circumstances, it made it very difficult to approach the mix with a clean slate. Our suggestion was for Allan to make future recordings with flat settings and minimal compression, at least until he got a feel for what could safely be done at the recording stage.

One of the first changes was to edit the Plate reverb setting being applied to the drums via the Lexicon MPX500, primarily by reducing the bass multiplier value from 1.0 to 0.6 and increasing the pre-delay to around 60ms. This took a lot of the muddy mid-range out of the reverb, resulting in a crisper drum sound.

The first sound we felt needed improving was the fretless bass, where Allan had used some chorus,

compression and EQ. The chorus sounded fine and the compression wasn't too far off, but the EQ he'd set up had a deep notch in the low mid-range, which meant there was plenty of deep bass but no real definition. Removing this dip and adding boost

centred at around 250Hz immediately reinforced the woody quality of the instrument. Once the rest of the mix was brought back in, it was obvious the bass sound was sitting much better. It sounded better defined and was punchy without sounding overblown — and it could still be heard clearly with the monitors turned down, when deep bass tends to disappear.

Still at the bass end, the kick drum was also lacking definition, something we remedied by adding quite a large amount of boost centred at around 4.5kHz and balancing this with an 80Hz hump. There was also some rather muddy ringing so we invoked the channel gate and adjusted the release time until we had what we felt was a tight, punchy kick sound.

This was the first time I'd used a Mackie D8b in anger and, though the EQ sounded very musical, it seemed to need far more cut or boost to get the job done than an equivalent analogue EQ.

Typically I might use 2-3dB of boost on an analogue EQ, but to fix this kick sound I found I was adding more like 10dB of boost! This is quite common with digital desk equalisers in my experience, although some of the later algorithms are a lot more analogue-like in this respect. Allan had used a dual-mic approach for the kick drum, with a D112 inside close to the beater head and a Superlux kick mic a little further out. Again, we had to EQ this second mic heavily in the 4-5kHz region to give it some attack, then we mixed it in with the main kick sound, but a few decibels lower in level, to provide a fuller and more rounded sound than the close mic could provide alone.

The snare had also been dual-miked with Shure SM57s, one mic above and another with the signal phase reversed on the bottom head. The upper mic needed some EQ boost at 6.1kHz to make it crisper, then the low head mic was boosted at 7.4kHz and brought up in level until we had a suitably convincing snare sound. Boost was also needed to freshen the hi-hats, in this case at 8kHz with a corresponding 140Hz high-pass filter to reduce the low-end leakage from other sources.

Acoustic & Electric Guitars

The 12-string acoustic guitar was peaked at 11.5kHz and dipped at 200Hz, the latter to cut boxiness

Some *Auto-Tune* Tips

Allan also mentioned that, although he had the *Auto-Tune* plug-in for the D8b, he'd never really got the hang of using it, so we called up a vocal track, established its key by playing along with a guitar, and set about adjusting the controls after setting the key input in *Auto-Tune*'s plug-in window. The most important control is the one that sets the speed at which the pitch is corrected, and once I'd figured out that this moved the opposite way from its VST plug-in counterpart (fastest correction when fully down), it was easy. The most natural correction was achieved by setting the correction speed slider about a third of the way from its top end (slowest position) where the pitch-correction display confirmed that corrections were only being made on sustained notes and that natural pitch variations were being allowed through intact. Allan was quite pleased to see this working so well as he had worried that it would sound unnatural.

from the sound, while a six-string acoustic was boosted at 10kHz to add shimmer. The compression Allan had applied to both these acoustic guitars was reduced until the gain reduction meters were showing no more than around 5dB of gain reduction, and

the attack time was set in the region of 20ms to allow the attack of the sound to come through clearly.

The track also featured a flanged electric guitar that was tending to clutter up the middle frequency region of the mix, so it was thinned by notching it at 150Hz and adding a 4.5dB boost at 2.2kHz. A high-pass filter was set at 50Hz to take out unwanted low end, then the guitar rebalanced in the track. A lead guitar that was also taking up a lot of space in the mix was similarly treated, but with a 210Hz notch and a 2.2kHz peak. The result was a powerful guitar sound that didn't stomp all over the vocals.

Moving back to the percussion tracks, the Tambourine was treated to heavy boost at 5.1kHz plus low shelving cut below 150Hz, and then the digital trim control had to be backed off by about 5dB because the HF boost was causing the channel to overload on peaks. A shaker was brightened by adding a relatively narrow boost at 9.5kHz. While all the EQ settings seemed quite drastic, they were needed only because that was the nature of the digital EQ being used.

The outcome of our endeavours was a much more transparent and open-sounding mix with better-defined drums and bass, complemented by lively acoustic guitars and percussion. It was brighter than Allan's original mix, but sounded more natural and airy, and no longer required the mastering EQ we had used to 'polish' the original version. The last job was to try to maximise the level of the mix so that it would sound loud when compared with commercial CDs, a job we attempted using the *Final Mix* software by setting the compressors to an infinite ratio and fast attack time to act as limiters, and also engaging the soft-clipping function. This bought us a noticeable amount of extra level, but wasn't as intuitive to set up, or as effective, as dedicated mastering software/hardware with a separate limiter.

Mixing Multitrack Drums

Allan also brought out a multitrack recording of an acoustic drum kit which sounded a little unnatural, partly because of the way the kit was tuned and played, but also because he'd EQ'd the toms to remove what he considered was excessive high end, presumably to minimise cymbal spill. Though most of the energy in a tom resides between 100Hz and 250Hz, the high end needs to be there to preserve stick definition, so we restored the missing high end and also beefed up the kick drum at 80Hz and 4kHz to give it depth and attack. Apparently the drummer wanted a ringy piccolo snare



Although the digital EQ in Allan's Mackie D8b mixing console sounded musical enough, much more drastic settings were required to solve his mix problems than would have been required using analogue EQ.

sound, but this wasn't really in the character of the drum. Nevertheless, by cranking up the mid-range boost, then sweeping it between 2kHz and 4kHz, we found a sweet spot that improved the sound. Once located, the boost was pulled back to a more practical level.

It proved to be impossible to gate the toms (to try to clean up the mix a little) because of excessive spill from the snare which caused false triggering. This seemed very odd given that the tom mics were apparently mounted just over the tom head rims, and implied that either the drummer didn't hit the toms very hard at all (unlikely!), or that they had been recorded with some compression in an attempt to create a fatter sound. Once again, this shows that it is often better to record the original signal raw, then apply gating and compression during the mix.

The kit had been recorded with stereo overhead mics, and Allan had taken the low end out of these during recording. All they needed was a bit of high-end encouragement to lift the cymbals

out, something easily achieved using boost in the 6-8kHz region. However, while this approach was fine for the intended purpose, it also meant that there was no possibility of using the overheads to form the basis of a complete kit sound, with the close mics then being added for extra definition, because the low end of the drums had effectively been EQ'd out. In other words, the mixing options had been severely limited with no real gains — another example of why it usually pays to record flat and with minimal compression, to keep all options open.

Allan had also gated the snare-drum mics quite hard, which affected the attack and resonance very audibly. The overall sound improved considerably when the gate range was set to around 10dB rather than infinity. This allowed a low level of spill to survive, which seemed to help the overall kit sound gel with the overheads, and it also improved the attack of the drums because some of the original transient could still be heard even though the gate took a short time to open fully.

In A Nutshell

The room problems we encountered were typical of those experienced by many project studio owners, and again we demonstrated that the

Allan's Session Notes



"I must admit that I was a little apprehensive about the visit, fearing that the guys might highlight too many imperfections in the setup, but far from it. I asked Paul and Hugh if they could concentrate on the acoustics of all the rooms and have a listen to some of my mixes to identify any fundamental mistakes I was making. A few changes to the positioning and settings of the monitors, addition of acoustic tiles and heavy drapes here and there and the room was sounding much better. More importantly, they suggested ways in which I could improve my mixes and, I have to say, since the visit, my clients and I have noticed a massive difference in the quality of the mixes."

worst of these problems could be improved at a relatively low cost. However, the control room still had a somewhat uneven bass response, largely due to its unfortunate geometry, so we suggested Allan fit two foam corner bass traps to each of the rear corners, in addition to finding a tidier and more permanent solution to the layer of foam and duvets we'd fixed to his front wall. We also felt that some one-metre-high speaker stands, ideally filled with sand, would tighten up the bass sound further.

Once the monitoring was sorted out, the reasons for the mixing problems became more evident, and it transpired that Allan had been perhaps too enthusiastic with his EQ and compression settings during recording, which had forced him into a bit of a corner when mixing. In a lot of cases, Allan had also been applying (almost as a habit) small amounts of lower mid-range cut at similar frequencies to try to reduce the perceived mid-range clutter in his mixes, rather than using the EQ to bring out the important elements of each source individually.

During mixing, we tended to use less compression than Allan had (over-compression can rob a sound of punch and clarity) and our final EQ settings were determined entirely by ear, simply by sweeping a parametric EQ across the frequency range while set at full boost to determine the best frequencies to cut or boost. Once these key frequencies have been identified, the cut/boost and bandwidth can also be adjusted by ear to give the best subjective sound. We also stressed the importance of making the final EQ adjustments with all the tracks playing, as what sounds great in isolation might sound quite wrong when the whole mix is up.

As a rule, EQ boost should be as broad as possible for a natural sound, though, in the case of drums and percussion, you can sometimes get a musically useful effect by making it narrower. EQ cuts on the other hand should be made as narrow as possible while still getting the job done, so that you're not taking anything more away from the sound than you need to.

On an artistic level, we also pushed the drums and vocals a little further forwards in the mix to get a more modern sound, while keeping any 'thickening' sounds further back in the mix to avoid congestion. A lot of modern mixes have most of their energy in the vocal, drums and bass parts, with everything else sitting lower in the mix, so choice of sound rather than sheer volume allows those parts to remain audible and distinct. **SOS**

Published in SOS July 2003

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SOUND ON SOUND

The World's Best Music Recording Magazine

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Studio SOS

Alistair Vickery and Jon Midwinter

Published in SOS September 2003

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People : Studio SOS

The SOS rescue team set off for Bristol to help out readers Alistair Vickery and Jon Midwinter, who wanted help with their mixes — not only getting the bass sounding right, but also stopping it getting out of the house and annoying the neighbours!

Paul White

Alistair Vickery and Jon Midwinter both started out as DJs, something they still find time to do, but they work together most of the time as a production duo in Bristol, specialising in drum & bass and recording under various guises, Distorted Minds being their primary tag. Alistair set up the studio six years ago and their first release 'Eventual', was rewarded with extensive club and radio play. Since then they have released material on various labels including New Identity Recordings, Formation, 5HQ Recordings, Renegade Hardware, Trouble On Vinyl and Breakbeat Culture. Today they run their own record label, D-Style Recordings.



Using the spectrum analyser within the channel equaliser in Emagic's *Logic v6*, the overemphasised low end of Alistair's mixes, and his bass sounds in particular, became very apparent.

The duo work from a large, single-room studio above a double garage at Alistair's parents' home. They called SOS because they felt their mixes were lacking clarity, and also to see if Technical Editor Hugh Robjohns and I could help with noise leakage problems. Like so many studios, their setup is a mix of hardware and software, with Emagic's *Logic Audio v6* providing the main recording environment. Monitoring is via a pair of Dynaudio BM15s fed from a

Hafler power amplifier, and most of the mixing is done manually using a Mackie analogue eight-buss console. John told us he wasn't convinced by most software instruments, as he found them too clean-sounding for drum & bass work, so many of the sounds come from a hardware Emu sampler and an Access Virus synth. There's also a rackmount Novation Bass Station and a couple of more conventional keyboard synths including a Roland JP8000 and XP50.

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Drum & Bass Mixing

The mixer tends to get used as a sound-design tool for reshaping sounds before use, often by applying radical EQ to individual sounds and then resampling them. We listened as Alistair played us some of his mixes and then some commercial recordings in a similar genre for comparison. Both Hugh and I felt that the sound from the BM15s was a bit bass-light, mainly because the dry-wall structure of the room was acting as a bass trap. A significant amount of bass was also escaping to annoy the neighbours, but other than an overall bass lightness there were no obvious hot or dead frequency spots.

Given that drum & bass isn't the cleanest of musical art forms, some lack of clarity is to be expected, but Alistair told us that when their material was mastered, it usually had to be brightened up quite a lot. The listening tests revealed that the bass end on their mixes was very deep and powerful, but it seemed to lack focus when compared with the commercial mixes, so rather than try to use EQ to correct the problem I asked Alistair and John to show us how they normally recorded and mixed their bass parts. The desk was already set up for the mix they were currently working on and, sure enough, on the bass parts the Low control on the Mackie EQ was turned up full to give +15dB of boost. Because this is a simple shelving equaliser, the boost continues all the way down the audio spectrum, so as well as beefing up the part of the bass spectrum that needed it, it was also pushing up the level of subsonic frequencies that simply eat into the available headroom without contributing to the mix in any useful way. This was almost certainly the main culprit in producing the muddy LF that Alistair and John had noticed when their tracks were played on big club PAs. However, because of the limited LF reach of the BM15s in their room, this subsonic energy was not apparent at all in the studio!

To see what effect this was having, I suggested we use the frequency analyser facility in the new *Logic* v6 Channel EQ to compare the frequency spectrum of their mix with that of a commercial mix they liked the sound of. The analyser needs to be set to its highest resolution to produce anything meaningful, and even then it can only provide a rough guide at the lower end of the spectrum, but straight away it was evident that Alistair and John's mix contained way more subsonic energy for more of the time than the reference mix. Hugh and I suggested that to improve this situation, they use a parametric rather than a shelving EQ to add any required bass boost, as that would avoid adding too much boost below the area of interest. Because only the largest professional monitoring systems cover the entire audio frequency range accurately (and then

only in a well-designed room), there will inevitably be things going on within a mix that you can't hear over a typical project-studio monitoring system. Furthermore, we pointed out that it's invariably better to create the right type of sound at source than to try to batter it into submission later using large amounts of EQ.

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Sorting Out A Rattling Speaker

When we started listening to the monitoring system to try to form an impression of the room acoustics, it quickly became apparent that something was amiss. There was a buzzy rattle that seemed to be triggered by certain bass frequencies. Tracking this kind of thing down can be difficult, as sometimes it's just as likely to be noises generated by other objects in the room, or even within the walls, as it is the speakers — especially if it only appears at high listening levels. So I had Paul play a few bass notes on a Roland JP8000 as test material to make finding the problem easier.

In this case we were able to establish that it was definitely a problem with one of the loudspeakers. Sometimes buzzy noises of this kind are caused by loose fixing screws, or even air leaks around the gaskets of the loudspeaker drivers, particularly if the speakers are routinely used at high levels and with bass/drum-heavy music. So the first and easiest thing to try is simply tightening up all the accessible fixing screws. In the case of the driver fixings, it is best to adopt a similar tightening procedure to a car wheel — tighten in small stages working around opposite pairs of screws. It's worth checking the fixings of the terminal connector panel on the back as well in the case of passive speakers, and the entire amplifier chassis in active ones.

In this case the driver fixing screws were, indeed, quite loose, but tightening them still failed to cure the problem. The next most likely scenario is that the internal connection wires, or even the lead wires running to the voice coil of the bass driver, are rubbing on something inside — other wires, sound absorption material, cabinet panels, or whatever. To solve this problem you usually have to carefully remove the bass driver and rearrange the internal wires as necessary — but if it is a lead-wire problem the bass driver will have to be replaced.

As we investigated, it turned out that the bass driver of the other speaker had already been replaced after



Hugh's first step when attempting to eliminate a rattle in one of the BM5 speakers was to secure some loose fixing screws on the bass driver. When this didn't solve the problem, Paul used a Roland JP8000 to provide a consistent signal, allowing Hugh to troubleshoot further.

exhibiting this very problem, and as it seemed likely that it was a similar lead-wire problem with this speaker, we didn't investigate any further, leaving it up to our hosts to organise a replacement as soon as possible. *Hugh Robjohns*

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Solving Low-end Problems

Alistair asked whether we thought adding a subwoofer would solve the problem, but we both felt that it would be inappropriate in this case. Firstly, to deliver very low frequencies with any pretensions of accuracy you need to have a very well-designed (and usually large) listening room, and a misleading and inaccurate low end would be likely to cause more problems, not less. Also, more extreme LF energy would pass through walls very easily, exacerbating the problem of escaping noise annoying the neighbours! The safest option in this situation is only to EQ those elements of the bass end you can actually hear and leave everything below 60Hz or so as it is. At least this will give the mastering engineer a fair chance of adjusting the bass end adequately before the record or CD is pressed.

We also ran a separate frequency analysis on the bass synth sound used in the track currently being worked on, as we all felt it lacked power and definition, even when extreme bass boost was being used. I suspected that the main reason was a lack of energy in the upper bass region where bass is heard rather than felt. This occupies roughly the 80Hz to 200Hz range, and it's mainly this part of a bass sound that you hear when a record is played on a small domestic hi-fi or on a portable radio. Sure enough, there was plenty of bass energy below 80Hz but a serious lack of activity for an octave or so above that, so trying to put the punch back with EQ wouldn't have helped the situation.



Alistair had been using fairly extreme EQ settings for his bass sounds on his Mackie console — the low-frequency shelf was cranked all the way up, giving 15dB of boost! Paul flattened the bass channel's EQ and resolved to optimise the sound in other ways.

One of the production tricks used by the pair was to dirty up sounds using the SPL Charisma valve processor, and in some situations they would split a signal into two mixer channels, EQ one channel to be very bassy and the other to be bass-light and bright, then add different amounts of distortion to the two parts (or distort just one of the parts) before mixing them back together. This can be a useful trick, but in this instance distortion didn't help, because although it added harmonics right across the spectrum, it did little to fill the 'hole' in the frequency plot. The only real solution when faced with this kind of challenge is either to modify the synth sound at source or, alternatively, to layer in another sound that contributes in the frequency region where the original sound is lacking. One

further recommendation we made was that Alistair and John should install a pair of small, domestic hi-fi speakers on which to try their mixes, as what sounds big on the Dynaudios can sound quite lacking in bass and punch on smaller speakers. Ideally, dance mixes of any type should be checked on studio monitors, small nearfields and a large club sound system to confirm that the bass end translates adequately.

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Using Distortion On Bass Sounds

The conversation turned to methods of distorting sounds, and I asked if they had ever tried using plug-ins. They said they had and that they didn't like the results, which didn't surprise me as I've never really liked them either, other than some of the better guitar amp simulators. I explained that raw distortion usually sounds bad, because it needs to be filtered by something like a guitar speaker or emulation of a guitar speaker to round off the rough edges. As the studio Mac wasn't a particularly fast model, I didn't suggest using a guitar preamp plug-in, but instead felt that a hardware unit (such as the Digitech Genesis 3 or the Line 6 Pod XT) might be a good solution for warming/dirtying up synths and samples, and it would also make guitar recording easier on the occasions they needed it. John told us that they have a good relationship with their local music store, and that it might be possible to try one out before deciding, and they definitely felt the idea had promise.

Purely out of interest, I set up some bass synth sounds when I returned home and tried using *Logic's* little-understood *Phase Distortion* plug-in to give them more attitude. Providing you spend a while experimenting with the controls (not all of which do what you'd expect), this can yield some very worthwhile results. The sound is hard to describe, but it's not unlike increasing the modulation depth on an FM synth, where the harmonics become richer and more intense. Although there's little in-depth detail on what goes on inside this plug-in, it seems to work by changing the waveform shape in a similar way to the phase distortion synthesis used in the old Casio CZ synth range (which was itself a form of FM) and can save the day when other forms of distortion fail. It seems to work best on monophonic synth lines and drum loops.

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Noise Reduction Measures

Before moving on to the noise leakage issue, Alistair told us that he had problems with the fan and drive noise from his Mac being picked up when recording vocals. It turned out that vocals were usually recorded to one side of the studio, but a few minutes of impromptu experimentation showed that a better place was towards the rear of the room, where the sloping ceiling/wall was covered in foam tiles. Working there only entailed buying a headphone extension cable, but setting up the mic facing the absorbent wall made the subjective level of the background noise at typical recording levels insignificant. The sound of the

vocals was also more natural and uncoloured when recorded from this position. A further improvement could be made by improvising an absorbent screen (the ubiquitous duvet?) between the vocalist and the mixing desk.

We also tried a trick I'd developed at home for making my own G4 quieter, which simply entails draping a heavy rug over it to form a tunnel, where the front and back of the computer 'look' out of either end of the tunnel. Keeping the back of the computer clear is essential, as this is where a G4 is ventilated. Providing the rug goes all the way to the floor on both sides of the machine, the drop in noise is significant, and can be further improved by having the back of the computer facing something absorbent, such as acoustic foam. We used a couple of heavy towels just to confirm the principle and everyone agreed the drop in noise levels was worthwhile.



A few experiments with some heavy towels confirmed that the noise of Alistair's Mac G4 computer could be significantly reduced with very little effort.

Prior to us finding this way to record vocals, the pair were planning to build a vocal booth, but in my experience homemade vocal booths are responsible for some of the worst vocal recordings I've heard. Typically they are treated to absorb well in the upper frequencies and mid-ranges, but not at the low end, so what you end up with is a middly, honky room, with a woolly, dominant bass end. If you have built such a booth and are having these problems, try introducing some reflective surfaces back into the room, for example wooden slats or diffusor/reflectors made from plastic guttering with the convex side facing into the room.

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Thoughts From Alistair & Jon

"The day Paul and Hugh spent with us certainly helped, and we're hoping we can try some of the sound isolation measures they described. I'm making more use of the EQ inside *Logic* now, especially with the analyser set to high resolution so that I can see the spectrum of the sound. I've also started to notice more about what we can actually hear on the BM15s (specifically at the low end) and I've tried out lots of different types of EQs to see what the differences are.

We're definitely looking at trying out a Pod XT, as some of our friends use one, and in general I think we had forgotten to experiment as much as we used to before the computer age really hit us. We bought the computer, got rid of almost everything else and then bought a lot of it back again! I think it's good to have both hardware and software tools and, although you can have a lot of fun messing around with either, at the end of the day you actually have to get on and write music. It also takes longer to realise what you can actually do with an application such as *Logic*, because there's an awful lot of stuff in there to explore."

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Dealing With LF Leakage

As expected, a lot of sound leakage was due to an ill-fitting outer door at the bottom of the stairwell leading up to the studio, and the single-glazed windows in the studio were leaking quite badly too, even though Alistair had made some Rockwool-filled shutters, which also excluded all daylight. In order to preserve the look of the building, we suggested fitting conventional double-glazed window units to the inside of the window spaces and leaving the existing leaded windows on the outside. One of the windows was right in front of the mixing desk, and would be reflective when the shutter was removed, but we felt the acoustics of the monitoring area could be maintained or even improved by fixing around one square metre of thick foam tiles to each side of the window opening, directly behind the monitors. As the side walls were quite a long way from the mixing position, there were no noticeable flutter echoes so it was decided not to treat these.



In an attempt to reduce the levels of bass leakage from the studio, Paul spent some time tapping various flat surfaces, such as underneath the stairs, to ascertain which ones were most likely to be resonating with the low frequencies.

The outside door was showing light around its edges, and this gap was allowing a lot of sound to escape around it — it was also not particularly heavy. We explained that, to make it an effective sound isolator, it should be made more massive by adding a layer of 3/4-inch chipboard to the inside, ideally with a layer of barrier mat (heavy, flexible, mineral-loaded material) sandwiched in between to add more mass and to increase the damping. The frame should then be extended to include a threshold strip at the bottom, and proper neoprene sealing strip should be fitted all the way round (not domestic draught excluder). These specialist items are available from any studio acoustics materials supplier. Modifying the door in this way may not be enough, but there was space to build a small entrance lobby with a second door, which would be much more effective should the need arise. Although it would be possible to make the outer door even more massive, a special frame and hinges would have to be fitted to support the weight, so our DIY solution felt like a good compromise.

Another major source of leakage was a lightweight studding partition wall at the side of the stairwell, which adjoined the garage. This allowed sound to pass out through and around the garage doors. Building an internal brick or block wall would be the ideal solution, but as there might not be space for this, we came up with a compromise plan which involved uprating the existing wall by adding a series of multi-material layers comprising barrier matt, 3/4-inch chipboard, soft

fibreboard and then one or two layers of thick plasterboard. Just applying this to the garage side of the wall should be adequate, but the same could also be applied inside the stairwell if needed. This is a relatively inexpensive solution, and within the scope of most people's DIY skills. The final plasterboard layer could then be skimmed with plaster to produce a professional finish. The reason for using this layered construction is to improve the damping of the wall, which in turn helps absorb sound energy rather than transmit it. The existing plasterboard wall could be felt to vibrate quite strongly when the monitors were playing at a moderate level.



"Can I borrow a cup of acoustic foam?"

Significant noise was also leaking from the underside of the stairs themselves and, again, a similar layered system could be used to damp this.

As the stairs were close to a garage window overlooking a neighbour, fitting internal double glazing to this window might also be of benefit. Some leakage through the studio floor into the garage was also evident, but curing this would be much more complicated, and could involve fitting a floating floor to the studio. As the other areas of leakage seemed the most significant, it was felt that tackling these would probably bring about an adequate improvement. **SOS**

Published in SOS September 2003

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SOUND ON SOUND

The World's Best Music Recording Magazine

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What Gordon Said

Thanks to Paul Eastwood of Audio Agency for supplying us with the Auralex acoustic foam components and adhesive.

Studio SOS

Gordon Giltrap

Published in SOS October 2003

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People : Studio SOS

The SOS team visit the West Midlands, where Gordon Giltrap's home studio needs help, nestling as it does in the shadow of one of the UK's most powerful TV and radio transmitters.

Paul White

Gordon Giltrap's unique acoustic guitar style has earned him a place amongst the world's top players and his career has taken many turns, from gigging with his own highly successful band in the '70s to touring either alone or with other genre-leading musicians, examples being Rick Sanders, violin player with Fairport Convention, or jazz guitarist Martin Taylor. Gordon also featured as the minstrel in Cliff Richard's *Heathcliff* musical and has a string of albums to his credit, but it's only in recent years that Gordon has taken the idea of recording his own music seriously. Because most of Gordon's compositions don't need much in the way of overdubs, he bought himself a Vestax HDR6 six-track hard disk

recorder, which also has in-built mixing capabilities. Teaming this with a small Mackie analogue mixer, a budget Lexicon MPX100 reverb and an AKG C414 microphone (now augmented by a pair of SE Electronics SE1s), he recorded and mixed a number of critically acclaimed albums, but on moving house he called us in to see if we could solve a few problems he was having.



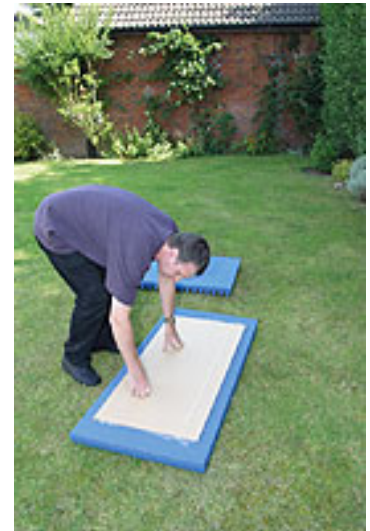
The nearby transmitter mast was spoiling Gordon's acoustic guitar recordings. However, experiments with kitchen foil showed that the problem could be solved by creating an earthed shield around the body of the guitar's pickup system.

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Eliminating Radio-frequency Interference

When Gordon told me he was suffering radio interference on his acoustic guitar pickup systems, and that he lived less than half a mile away from one of the most powerful radio and TV transmission masts in the UK, I thought we were going to have our hands full. On top of this, the room/monitor setup wasn't working as well as it should, and the whole system was afflicted by a low level of hum. Furthermore, because the room (a former bedroom) was acoustically untreated, there were reflection problems colouring both the monitoring environment and the miked sound of the guitar. We felt this might require two visits, one to assess the problems and another to bring together all the solutions.





Here you can see Paul building a couple of MDF-backed acoustic foam panels to tame room problems in Gordon's studio. First the MDF was sawn to size and sanded down, before having holes drilled in the corners, through which string was tied to allow wall mounting. After a brief glance at the instructions on the can, spray adhesive was applied to both MDF and foam, and the surfaces finally pressed firmly together.

Upon our arrival, Gordon demonstrated the radio pickup problem using one of his acoustic guitars fitted with a Fishman Rare Earth pickup system comprising a magnetic pickup and a small mic with integral preamp. Radio signals are of course far too high in frequency to be audible, but as soon as they encounter any circuitry with semiconductors, the radio signal gets rectified, and if the signal is from an AM transmitter then the programme material becomes audible. Although FM transmissions are harder to accidentally convert into meaningful audio, and digital signals all but impossible, they can still produce noise-like signals in the audio band. The only real cure is either to saw down the transmitter (hardly practical in this case!) or to remove the offending radio-frequency signals before they can reach any active circuitry.

Small parallel capacitors or ferrite sleeves fitted over cables can be very effective in attenuating radio-frequency signals without significantly affecting the wanted

audio, though there are only around four or five octaves separating the audio and the carrier frequency of some of the long-wave radio stations, so something sharper than a 6dB/octave passive RC (Resistor/Capacitor) filter might be needed in some situations.

After trying clip-on ferrite chokes on the guitar lead and on the cable inside the guitar joining the pickup to the jack socket, it became clear that very little improvement was being made. This really didn't surprise me, as I felt the RF was most likely being picked up in the windings of the magnetic pickup and then being rectified by the pickup preamp circuitry, which is housed in the same physical unit. The other way to remove RF is to prevent it getting to vulnerable parts in the first place, something that can be achieved by placing a conductive screen around the part and then grounding it. To verify this experimentally, I wound aluminium baking foil around the pickup assembly leaving the top open (the part facing the strings) and then used an ordinary jack lead to temporarily connect the foil to the ground of the guitar's output jack, simply by holding the cable in place. The result was better than we could have hoped for, and the phantom radio programme in the background was silenced.



Hugh installs the newly made wall panels, using a spirit level for a neat result, while Paul carefully sticks acoustic foam bass traps directly into the front corners of the room using more spray adhesive.

Of course baking foil is not really a permanent solution, because even if you could fix it in place using double-sided adhesive tape, it's very difficult to make a reliable electrical connection to it and it can only be soldered using special flux. The more professional solution is to use either conductive paint or adhesive-backed copper foil, the latter being my preference, as you can solder directly to it. This could be linked via a thin wire to the ground side of the battery compartment or to the ground of the output cable. As several of Gordon's guitars would need to be modified in this way, we suggested that he get his usual guitar technician to do the job for him based on our solution.

Gordon subsequently ordered some adhesive foil and modified one of the pickups himself and found that the screening worked OK, but that if he grounded it to the battery compartment the battery became permanently live and so the battery drained very quickly. To do the job properly would involve running a wire to the 'earthy' side of the output jack, but as Gordon was now getting good results using only microphones, he hadn't got around to fixing this by the time of our second visit.

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Solving The Hum Problem

The hum problem turned out to be a ground loop caused by the use of unbalanced jack cables between the Mackie mixer and the power amplifier. In fact all the system connections were made using unbalanced cables, so we suggested replacing these with balanced cables. The Vestax manual didn't say whether the outputs were balanced or not, so we assumed they were not. Unbalanced leads can be used to connect to a mixer in this case, but where the source is unbalanced, it's better to use specially wired unbalanced-to-balanced cables, as these reduce the risk of ground loops. The strategy is simple — wire the balanced end of the cable to a balanced jack in the normal way, but at the other end of the cable use an unbalanced jack with the hot conductor connected as normal, the cold connected to the tag on the jack plug where the screen normally goes, and with the screen left disconnected.

This same connection system could also be used to connect Gordon's Roland GR33 guitar synth or his Kawai K1R synth module to the mixer, but, as a rule, he patches these directly to the inputs of the Vestax machine when recording. Conventional balanced cable could be used to connect the Lexicon MPX100 effects unit to the Mackie's aux send and stereo return, and also to connect the mixer's control room outputs to the rackmount power amplifier driving the speakers.

On our return trip Gordon had replaced all his system's unbalanced cables with balanced ones. He hadn't had any balanced-to-unbalanced cables made up, but the new cables completely cured the hum problem and worked fine with the Vestax machine, so perhaps that had been balanced after all!

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Rearranging The Studio

The next step was to look at the arrangement of equipment within the room. Because of a centrally placed radiator beneath the window, Gordon had decided to set up his equipment along a side wall, but this meant the setup was acoustically asymmetrical, with the window to his left and a solid wall with a door to his right. The monitors were perched on fairly unstable tall CD racks right in the corners, which usually leads to an inaccurate and unpredictable bass response, and, as there were no curtains fitted at the time, there were significant flutter echoes between the window and the opposite wall.

I felt that the gear could be moved round so that the mixer would be below the window, in front of the radiator — this was unlikely to be turned up very high when the studio was in use, because of the heat generated by the equipment. As a precaution, foil backed foam of the type normally put behind radiators to reduce heat loss through the wall could be hung over the front of the radiator to reduce the amount of heat radiated towards the equipment when the radiator was on. We recommended that the speakers be placed on solid stands rather than the CD racks and be moved a little way inwards from the corners of the room to minimise

bass problems.

With the studio equipment arranged symmetrically around the window, flutter echo problems could be addressed using acoustic foam, and once again Paul Eastwood of the Audio Agency managed to find us a quantity of blue Auralex panels, plus a couple of sets of foam corner bass traps in purple, which would go nicely with Gordon's room. I was a little worried about sticking panels directly to the wall, just in case Gordon decided to move again in the near future, so instead suggested that we hang the panels on the rear wall and to each side of the listening position after first fixing them to MDF or hardboard using spray adhesive. This would allow the panels to be hung from nails, just like pictures. We had to fix the bass traps (which had to go in the front corners, as the door at the rear of the room was right in one corner) more permanently using spray adhesive, but Gordon was happy about doing this.



Without the excess reflections which were colouring the room sound, the guitar sound benefited from placing a reflective surface beneath the guitar, adding some lively early reflections.

Gordon had been mixing his tracks using the internal mixer in the Vestax, and though this can be driven from an external MIDI hardware controller such as a Kenton Spin Doctor, Gordon had been using the menu buttons and LCD window — not very hands-on! However, there were just enough inputs on the Mackie mixer to accommodate all six direct outs from the Vestax, which we felt would be easier and just as good sonically (if not better), so rather than set up a hardware controller, which had been my first thought, we went this route instead.

Gordon was also interested in getting hold of a simple, relatively inexpensive mic preamp/front end and eventually settled on an SPL Gold Mic, which he had installed by the time of our second visit — he says he is extremely pleased with it.

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Some Of Gordon's Recordings

Gordon has now recorded several CDs on his home setup, all of which are available on Voiceprint records, including:

- *Janschology* (six-track tribute to Bert Jansch), LCVP124
- *Troubadour*, LCVP147
- *Under This Blue Sky*, LCVP150
- *Remember This*, LCVP155

 www.giltrap.co.uk

The Return Visit

By the time of our second visit, Gordon had already moved the room contents round by 90 degrees and bought proper Atacama speaker stands. The speaker stands employ two thin vertical rear tubes and a larger-diameter front tube between the base plate and top plate, and the front tube was quite resonant. To cure this potential ring we suggested filling the tube with dry 'playbox' sand, which would mean unbolting the top or bottom plate and filling the tube, using the cut-off top of a plastic drinks bottle as a funnel.

The base plate was fitted with four adjustable spikes, but these were very difficult to adjust — the result being that the tall speaker stands tended to rock slightly on the carpet. We suggested buying a couple of small concrete garden slabs and using those to provide a firm high-mass base on the carpet, placing and levelling the stands on top of the flat surface. The speakers could be secured in place on top of the stands with Blu-Tac or similar.

Gordon also had some sheets of quarter-inch MDF prepared, to which we could stick the foam panels, so we were pretty much set to go. However, because of the furniture in the room and the fact that some of the wall space was being used to hang guitars and pictures, we reasoned that a single four-foot by two-foot panel on each side, and one on the rear wall, would be sufficient. This meant cutting one of his MDF panels in half (we originally anticipated using four half-metre tiles per side), after which we drilled holes in them, through which we could pass string or wire for hanging them up, rather in the style of paintings. The string was fitted to the panels first, then the foam was glued into place using the spray adhesive provided (Note, I did read the instructions on the can first!). Around 3 inches of foam protruded from each edge of the MDF sheet so the MDF was quite invisible once the panels were hung in place. The foam bass traps were glued directly to the walls using the same adhesive, and this process takes a little care as the adhesive grabs on contact — you definitely need to line up the traps carefully before pushing them into place!

Once the Auralex foam was fitted, we played some of Gordon's CDs and immediately discovered a tighter, more focused sound that was more even at the bass end. As the room was carpeted, we suggested using the spare sheet of



The original setup (above), along the side wall, and the final arrangement (left), shifted to the front wall for better acoustic symmetry.

MDF to place on the floor to add a few reflections while recording the guitar with microphones, and Gordon tested this by playing a few songs with the MDF beneath his chair while listening carefully to the tone of the instrument. As expected, the guitar sounded a little more lively with the MDF in place and, because the Auralex foam had tamed the worst of the room's coloration, the acoustic sound of the instrument was much more pleasing than before the room was treated.

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Summary

As with many of our previous visits, what was initially described as one problem actually turned out to be several smaller problems working together. The hum and interference problem was cured in the main system by using balanced connections between balanced pieces of equipment, while the monitoring was improved by using proper stands, moving the speakers away from the corners and fitting some strategically positioned acoustic foam.

A bedroom 12 feet square is never going to be transformed in to an acoustically perfect recording and mixing room, but the application of a modest amount of acoustic foam to the sides and rear of the room plus the fitting of four corner bass traps improved the situation very significantly. The window, of course, remains reflective at mid-range and high frequencies, but Gordon is planning to fit vertical blinds, which can help break up reflections to a worthwhile extent when they are set to their half-open position. The sum of all these small improvements transformed a badly behaving room with noisy equipment into one in which Gordon should be able to continue to make high-quality guitar recordings, free from electrical hum, radio interference or excessive room coloration. **SOS**

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What Gordon Said

"Having worked in some of the best and worst recording environments imaginable, I have some idea of what a good-sounding room should sound like, but never would have imagined that my converted spare bedroom would end up sounding so good to these tired old ears! The main impression is one of a warmth that definitely wasn't there before — the treatment has pulled the whole thing together into a much more controllable and focused sound.

"The difference, once Paul and Hugh had completed the exact placement of the tiles was truly remarkable. Not only have they transformed that room into a proper recording environment, but the tiles look great to boot, almost a work of modern art in themselves. I can't wait to start work on the next batch of recordings to get the full benefit of this sonic makeover!



"I feel fortunate in having Paul White as a friend and for being able to call upon his wealth of knowledge when it come to all things of a recorded nature. I was also very pleased to meet Hugh at last. Now all I have to do is buy sand and decorative slabs for the speaker stands — because they'll probably come back and check up on me!"

Published in SOS October 2003

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Thanks to Paul Eastwood of Audio Agency for supplying us with the Auralex acoustic foam components and adhesive.

Studio SOS

The Harding Family

Published in SOS November 2003

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People : Studio SOS

Musicality isn't everything when you're trying to make a decent-sounding recording at home, even if you're as talented as the multi-instrumentalist Harding family. So the SOS team travelled north to help them get their engineering techniques up to scratch.

Paul White

Recently SOS's Sam Inglis attended a function at which the entertainment was provided by a family band comprising parents and their three daughters. Normally this type of line-up, especially when all three girls are still at school, is a recipe for hopelessly cheesy music and sympathy-based applause, but Sam (who tends to be very critical of live performances) returned full of praise both for the playing abilities of the band and their songwriting abilities. In a subsequent conversation with them, he discovered that they had bought a Yamaha AW16G recorder/mixer, a pair of Samson active monitors and three Shure SM58 microphones to set up their own studio at home. Furthermore, they even provided him with a CD of work in progress, the quality of which was surprisingly good and which would have put many a Demo Doctor entry to shame! However, the drum sound wasn't up to scratch, even though the playing most definitely was, and the overall balance and tonality of the mix could have been better, so Sam handed the CD over to myself and Hugh so



In order to treat a marked room resonance at 110Hz, each speaker was first isolated from the desk, which was exacerbating the problem, using two heavy quarry tiles separated by the speakers' rubber feet. A further improvement was made by inserting a small piece of acoustic foam into each speaker's reflex port, raising the low-frequency roll-off point.

that we could arrange a visit.

The Harding family band play a wide variety of styles of music including folk, jazz, rock, and pop. They have performed widely, as well as appearing on both local and national radio — they were even on BBC Radio Four in 1996 when they played for, and were interviewed by, John Peel. Tim Harding is a primary school head teacher and his wife Amanda is an archivist now working in schools presenting Living History workshops. They have been playing music together with their three daughters — Emily, Charlotte and Eleanor — since the girls were old enough to pick up an instrument, and the girls now play over 14 different instruments between them. Emily (aged 16) plays drums, flute, keyboards, percussion, violin, and sings backing vocals. Charlotte (aged 14) plays alto/soprano saxophones, clarinet, whistles and keyboards, and takes the lead vocal in most of the songs. Eleanor (aged 12) plays trumpet, violin, mandolin, whistles and keyboards, and also provides lead and backing vocals. Tim adds bass guitar, guitar, mandolin and trombone, while Amanda plays keyboards and violin. Emily and Charlotte currently write most of the material, and like experimenting with different styles of music.

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Studio Treatment

Situated in a 17th-century house in a small village south of Kingston-upon-Hull, the band's studio comprised an extremely small control room (2.45 x 1.8m) adjoining a 4 x 4m live room that also doubles as a rehearsal space. Hugh and I were instantly suspicious of the monitoring accuracy of such a small control room, especially as there was no acoustic treatment at all, so while we partook of our coffee (and the obligatory chocolate biscuit!) we asked to hear a playback of some work in progress. Immediately our suspicions were confirmed, as the room obviously had a nasty resonance at around 110Hz that made the low frequencies very uneven, with almost a 'one-note' bass characteristic, tending to make mixes sound boomy. We agreed that we had to try to improve this before tackling any of the other problems, so we did some more listening using a couple of commercial recordings just to confirm the boom was a room problem and not a recording issue. It turned out most definitely to be a room problem.

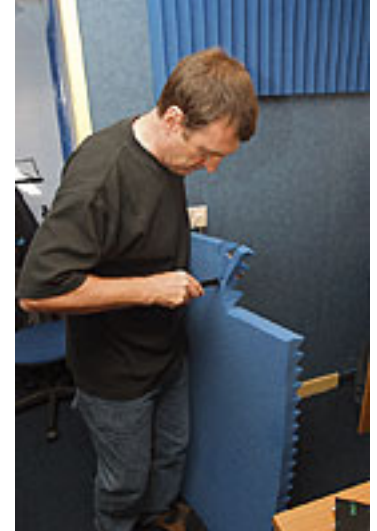
Ultimately, we had to apply several remedies, each of which made an incremental improvement to the monitoring accuracy of the room. As the rear wall of the room was almost directly behind the mixing chair, we decided to treat this with a couple of panels of Auralex foam that we had brought with us. This necessitated removing a few hooks and fittings that had been used to store cables on the rear wall, and we also had to cut away the corner of one of the panels (using a serrated kitchen knife) to clear a mains socket. The panels were fixed with spray adhesive and perfectly matched the blue decor of the room.

We were also worried that having the speakers standing directly on a lightweight desk might cause resonance problems, so after confirming that the desk did indeed resonate at around the offending 110Hz when tapped, we decided to try

raising the speakers off the desk by a few inches, first by improvising supports out of box files, then more permanently by using some thick clay quarry tiles left over from renovation work on the house. These Samson monitors come with detachable rubber feet, so we used two quarry tiles under each speaker with three of the rubber feet between the two tiles to act as an isolation mount. This turned out to work pretty well, though we recommended placing some non-slip matting beneath the speakers as the tiles (what with being handmade and pretty ancient!) weren't entirely flat.

While moving the speakers, we also noticed that the mid-range frequency trims had been set to -3dB, so we reset them to flat and played our test records again. Sure enough, the 110Hz honk was less noticeable, but it was still there. I had an idea that the space beneath the desk was acting as a resonator and suggested packing some sleeping bags (one of the girls also suggested soft drum cases) under there as an experiment. We also tried putting some rubber car mats on the desktop to damp it. Again we heard a slight improvement, but it wasn't a complete cure, so another recommendation we didn't have time to try at the time was to cut holes in the desk sides so that it wouldn't behave quite so much like a tuned box. We felt it would also be worth trying to stiffen the desktop from below using something like 3 x 2-inch wooden braces.

Our final experiment turned out to be one of the most worthwhile. To reduce the degree to which the speakers were exciting the remaining room resonance, I pushed some scrap pieces of foam into the speaker bass port slots, the idea being to lower the Q value of the cabinets and start the bass end rolling off at a slightly higher frequency. This simple and easily reversible fix tightened up the bass considerably and, though the bass level in the room still wasn't perfectly even, individual bass notes could now be heard quite clearly and they weren't



After removing some cable storage hooks from the wall behind the monitoring position, some Auralex acoustic foam was attached to attempt to improve the acoustics in the control room. One corner of the foam needed to be customised, using a kitchen knife, to fit around an existing mains wall socket.

being obscured by the 110Hz honk. Given the small room size, we felt this was about as far as we could go in improving the monitoring environment, and everyone felt the sound was significantly tighter and more accurate.

During these tests we'd also discovered a low-level, but nevertheless annoying, hum on the monitors that didn't respond to any of the usual ground loop fixes (such as lifting screens). In the end, Hugh discovered that this was not caused by a ground loop at all, but that it was induced from a Korg X5D keyboard wall-wart power supply which was lying adjacent to the mains cables of the speakers and hard disk recorder. By moving the wall-wart away from these cables the hum disappeared, so we ended up simply plugging the wall-wart into another wall power socket, whereupon the hum was completely eliminated.

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Sorting Out The Drums

The drum kit turned out to be a rather nice Premier kit, fitted with decent heads and pretty well tuned. The band had been recording the kit using just their three SM58s, one on kick, one on the snare and one as an overhead. Their demo recording had a particularly tubby kick drum sound, and though the SM58 isn't an ideal kick drum mic, due to its lack of extended bass response, the real problem was traced to the way the drum was tuned and damped. A friendly drummer had recommended damping the drum by filling it with yards of bubble wrap, but it didn't seem to be doing nearly such a good job at damping as the traditional folded blanket! Unfortunately, the front drum head had only a small hole cut in it (one of the less fortunate current fashions as far as recording is concerned) which meant there was insufficient room to get a blanket into the drum and no way to get the mic into the proper position, so we had to resort to removing the front head and working without it. My preferred option is to use a front head with a large hole cut in it, as that not only provides better access, but also leaves less of the head to resonate — something that can lead to a messy or flabby kick sound. Leaving the front head off in the long term is not recommended, as lesser drum shells may distort under the uneven tension, although I've never known this to happen. A more likely problem is that



The band had been working with their bass drum filled with bubble wrap, on the advice of another drummer. However, this was not producing a suitable sound for recording purposes,

you may lose the fittings, or the tensioner nut boxes may rattle — as it was, we only had to spray the pedal with WD40 to cure a squeak.

so it was removed and replaced with some heavy blankets. The front head was also removed, because the hole cut into it was too small to allow much flexibility in mic positioning.

Removing the head and putting a couple of folded blankets into the drum tightened up the sound, but I still felt the drum was tuned a little too high, so I dropped the pitch slightly, which gave a more satisfying splat at the attack of the note. The pedal was fitted with a hard felt beater, but for recording a solid wood or plastic beater usually gives a better tone, so we improvised by sticking a plastic ID card to the head using gaffer tape, positioned so that the beater hit the centre of the plastic card. Everyone agreed that this was much more like the kick sound they were after, as it had both depth and definition.



Even when the front head had been removed and the bubble wrap replaced with blankets, the sound still needed more attack and less ringing. A little tissue gaffered to the skin and an ID card taped over the beater contact point helped here.

The other drums were tuned pretty well, but needed a little more damping, which was provided using small amounts of folded tissue held in place using masking tape or black gaffer tape. (Which wise man was it who said that gaffer tape is like 'the force' in Star Wars? It has a light side, a dark side and it holds the universe together!) It is very important to damp the lower tom heads slightly, as these will ring whenever the kick drum is played, and in most instances this ring will be picked up and recorded, making the kit sound boomy and ill-controlled. Although this can be cured to some extent by the use of gates, setting up can be tricky, and it's easy to make matters worse rather than better.

As luck would have it, I had brought a set of Samson drum mics with me, so I set them up on the kit, close-miking each individual drum and using the two included capacitor mics as overheads. Because the AW16G has a limited number of mic inputs, and because only two of those are on XLRs, we used the band's Peavey XR800F powered mixer as a submixer for the drum mics, and for the purposes of

our tests these were fed directly to two tracks on the recorder. However, for maximum flexibility when mixing, it would be better to use two more tracks to enable the kick and snare mics to be kept separate, as this allows the snare reverb to be controlled independently and also makes it possible to adjust the kick and snare balance when the rest of the mix is up and running.

With just a little mid-range EQ cut on most of the close mics, we ended up with a more than respectable drum sound, and the best results were achieved when we used the overhead mics as the basis for the sound, then brought up the close mics just enough to create a good balance between the drums and cymbals. Using the overheads to provide the main body of the sound also lessens the effects of ringing drum heads, as these are picked up mainly by the close mics.

Close mics on their own can sound a little dull and ponderous, while in this case the overheads alone were bright and lively, but noticeably cymbal-heavy. We positioned the overheads slightly towards the rear of the kit to reduce the volume of cymbals that they picked up. Getting the ideal balance required making several test recordings, then listening to the playback, as the acoustic isolation provided by the light, ill-fitting 'historic' door between the control room and live room was more psychological than real! I didn't mention neoprene seals and heavy fire doors as I didn't think this was the feel they wanted...



A pair of condenser mics used as stereo overheads really helped give a more professional sound. Positioning them a little further back gave a less cymbal-heavy balance, which meant that only a little of the each close mic was required to focus the sound of the drums.

Having proven that we could get a good drum sound, we changed the rig to use two of the band's own SM58s (kick and snare) plus a single Samson C03 budget capacitor mic in cardioid mode (with the -10dB pad and high-pass filter switched in) as a mono overhead. We didn't use any close tom mics in this instance. Our overhead mic was pointed roughly at the drummer's lap so as to avoid picking up too much cymbal and to give the toms a chance. The reason for using a single overhead mic and for bringing back the SM58s was that the band said they could afford to buy one capacitor mic for recording vocals, brass/woodwind and drum overheads, so we wanted to see what results could be achieved that way. Obviously, you lose the stereo feel of the drum kit, but a little stereo reverb can help restore some of the missing spatial ambience.

After a few minutes getting a good balance, we ended up with an acceptable drum sound that had good definition and tone, but where the toms were a little quieter than ideal and the kick drum lacked the depth of the dedicated kick mic we'd used earlier. Nevertheless it wasn't bad at all, and was a great improvement on the band's original recorded drum sound.

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Sax & Vocal Miking

For live performance, the sax is currently miked using a small clip-on mic fixed to the bell of the instrument, though studio recordings (always made as separate overdubs) had been captured using one of the SM58s pointing at the bell of the instrument. The sound wasn't bad, but it lacked definition, and some of the more subtle nuances of the sound had been lost. We hung up as many blankets, duvets and sleeping bags as we could find in one corner of the live room to provide a non-reflective backdrop, then used the Samson C03 mic (again in cardioid mode) around 14 inches from the body of the sax and aimed at the body of the instrument a couple of inches or so above the bell. With no EQ or further

processing, this immediately produced a nicely moody sound that Lisa Simpson would have been proud of, with lots of air and definition combined with the warmth. A hint of reverb and we were all happy with the result. Given that the Samson mic is around the same UK price as an SM58, this was quite remarkable.

A similar setup was used to record the vocals, again with the mic pointing towards a duvet-treated corner. Previously the vocals had been recorded in the control room, with all its resonance problems, using an SM58 and a pop shield. Actually the results had been pretty good, but Hugh and I felt that we could improve on the clarity and also lose some of the room coloration by using a capacitor mic (again plus pop shield) in the larger live room. A test recording using the C03 confirmed that the vocals did indeed seem more lively that way, and again all that was needed was some compression and a sympathetic reverb to make them sound as though they belonged on a serious record. No EQ was really necessary for either the vocal or sax recording made in this way, though a little air EQ is useful if the high end needs flattering.



Once a successful sound had been achieved with stereo overheads and individual drum mics, Hugh set about achieving a comparable sound with a more modest setup of two dynamic mics and one condenser.

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Effects Advice

Tim Harding admitted that he had little experience adjusting effect and processor settings, and so tended to rely on the Yamaha AW16G's presets. In our experience, presets often don't give the best results, so we created a few custom patches and explained what we were doing as we went along. Calling up a vocal compression preset is fine, but compressors only do their thing when the input exceeds a threshold, so they behave differently depending on how loud you record the track in the first place. If, for example, a singer is consistently loud and close to peaking all the way through a song, the compressor will work nearly all the time, whereas if the track was recorded with the peaks at around -12dBFS or so, the compressor might never kick in, even though the compressor settings are exactly the same in both cases. If you must use a compressor preset, then at least adjust the threshold control while listening and watching the gain-reduction display.

Normally the compressor should show little or no gain reduction on the quieter phrases and around 8dB maximum on the louder phrases, though you may need to compress a little more than this if the singer has a very wide dynamic range. Ideally you should compress just enough to get the vocals to sit evenly in the track. I find that a ratio setting of 4:1 is enough for most singers and that an

attack time of between 5ms and 30ms with a release of around 0.25s usually works well. I tend not to use any vocal EQ unless the recording obviously needs it, and as most voices are different it's difficult to see the point of vocal EQ presets. Having said that, you may find the ubiquitous 'air' EQ trick of applying a small amount of broadband boost at 12-14kHz helps to add clarity and presence to a vocal that needs it, especially if you are using a mic with a less-than-sparkling top end.

I also set up a couple of custom vocal reverb settings, plus a drum plate. These used the presets as a starting point, and my first vocal treatment was based around a vocal plate: I added brightness and reduced the bass multiplier value to below unity to get sizzle without muddiness. The reverb time can be anything from 1.8s to 2.5s, depending on what the song needs, and I used a pre-delay of 70-80ms to create a sense of space. The second vocal patch was based on the Live algorithm, with around 40ms of pre-delay and again some extra top end. This patch also sounded very neat on sax.

For drums I chose a bright plate, and combined a delay time of around 1.8s with a shorter pre-delay — any more than around 40ms can sound odd on drums, unless you specifically want a doubling effect, and as a rule the shorter the reverb time, the more reverb level you can add and still get a natural sound. In most instances, you only need to add reverb to the snare mic, with a little also on the overhead(s), just to help knit the sound together and to take the dry edge off the hi-hats.

My final foray into effects tweaking was to set up a very gentle compressor that could be used in the master stereo mix insert point, and also a 'smile' EQ setting that could add a little sizzle and depth to a finished mix. This is no substitute for professional mastering, but can help give finished tracks a more polished sound. If you're having a track professionally mastered, though, it's best not to apply any global effects of this kind to the mix. Our compressor setting used a ratio of 1.1:1, with a threshold setting of -40dB combined with fairly fast attack and release times to gently compress the whole



Charlotte's vocal and sax sounds had been suffering from being recorded using an SM58 in the less-than-ideal acoustic environment of the control room. Using a comparatively low-cost condenser mic in the live room, with drapes and duvets damping the corner behind the performer, immediately offered a much more usable sound in both cases — hardly any EQ was necessary once these measures had been taken, although a pop shield was needed in both cases.

of the mix — more conventional compression only treats the peaks, but it does so much more assertively.

Our 'smile' EQ simply combined gentle bass boost in the 80Hz region with a wide and shallow mid-range cut at around 250Hz plus 'air' EQ boost up at 12-14kHz. This should be subtle enough so that you are just aware of the extra clarity when the EQ is switched in. It's hard to put precise cut and boost figures on this, as an analogue EQ may achieve with one or two decibels of cut and boost what some digital EQs need 7-8dB to achieve. In the case of the AW16G, we needed 4-5dB of adjustment at the high end to get the requisite amount of air into the mix, while around 3dB of adjustment was generally enough in the mid-range and low areas.

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Comments From The Hardings

"This was a great day where we all learned a lot. Thanks to Paul and Hugh's excellent DIY skills, the control room is already providing a much better sound environment for monitoring recordings — the difference made by a few relatively simple adjustments was very noticeable indeed! Our number one priority for upgrading will be to buy one or more condenser microphones and, as we're funding a whole band here, we initially need a microphone that is adaptable for use on voices, a range of acoustic instruments, and drum overheads, as well as being affordable. Something like the Samson CO3 or one of the other low-cost cardioid condenser mics will be top of the list, probably followed by a set of drum mics, as budget allows.

"We'd had no real previous experience of miking drums up at home, so it was great being shown how to get a convincing sound. We'd never realised that gaffer tape, masking tape and tissues would make such a big difference and Emily was particularly pleased with her tom sounds once Paul had doctored the kit. Using the clip-on drum mics was fun, and it was amazing how much clearer the drums sounded. However, even when the drum mics were packed away, Paul and Hugh still managed to get a very reasonable sound using the mics we already had, the best feature of which was the plate reverb effect, which gave the kit a really live feel.



"The biggest change to the kit though, was the front bass drum head being removed. This coupled with more gaffer tape, tissue, tuning, and an ID card taped to the head gave the bass drum a fantastic, punchy sound. This gave us the sound and feel the songs needed, without Emily having to wear her leg out!

Although Emily might stick with the bubble wrap for live performances (she really does like the sound!), we'll definitely be using blankets for recording, and we have already altered a mic stand specifically for use in the kick so the mic can get right inside the shell. The overhead condenser mic gave a real sparkle to the cymbals, which gave the whole sound more fizz.


"Normally, we've recorded the sax parts in the small studio, just with an SM58 hovering over the bell. Moving the mic away from the bell just a little bit, and aiming it more at the body of the sax definitely picked up a much rounder, fuller tone. The condenser mic also helped to define and clarify the sound, while placing quilts and curtains on the wall improved the quality of recording no end. The obligatory reverb treatment enhanced the whole sound and Charlotte was extremely pleased with the end result!

"Again, Charlotte would normally have recorded her vocals in the smaller room with an SM58, but as we found with the sax, working in the big room against the absorbent backdrop brought what the mic was picking up into focus, and again the condenser gave that bit of extra clarity. Paul's EQ and dynamics settings were very helpful and I'm sure they'll be used a lot! One of the things we'd wondered about was whether to add reverb while recording or add it later. Charlotte's vocals were better when she'd sung with reverb in the headphones, so we were glad to discover that she could have reverb in the phones when recording, but not actually record the reverb. Adding it later means we'll be able to experiment with the different settings more.

"The advice on effects and processors was particularly useful. Tim felt that there was obviously much to be gained from experimentation, although he also felt that the key to this was confidence and, already since the visit, he's investigated various effects based on Paul's guidance and everyone's noticed the difference. Particularly satisfying was the stereo 'smile' EQ setting which added a bit of magic to the overall sound."

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A Bit Of Everything

Because there were so many separate issues to address, we felt that we hadn't gone into anything in great detail or depth, yet the improvements we'd made to both the monitoring environment and the miking of drums, vocals and sax were significant. The control room boom had been tamed to manageable proportions, the Auralex foam had tightened up the imaging by cutting down rear-wall reflections and the *ad hoc* speaker mounts and foam plugs had improved the evenness of the bass end to a worthwhile extent. I felt that we'd also demonstrated the benefits that even budget capacitor mics (combined with strategically placed duvets, of course!) can bring, and we even managed to throw in a bit of drum tuning and effects tuition! 

Published in SOS November 2003

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Studio SOS

Clive John

Published in SOS December 2003

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People : Studio SOS

Recording piano was turning out to be a grand challenge in Clive John's home studio, so the SOS team headed over to sort things out.

Paul White

Clive John's studio holds no shortage of electronic instruments (some quite old and desirable!) and computer software, but the focus of this particular challenge was Clive's fabulous Fazioli grand piano, which he was having difficulty recording. The main problem was thought to be the room, which was roughly square and probably no more than five meters along each side. Combined with a low ceiling (around 2.3m), this was giving the piano a somewhat boxy, coloured sound that did no justice to its true capabilities. The floor was quarry-tiled, and the outside wall had a window and door.

Clive had experimented using a couple of duvets to try to deaden parts of the room that he knew to be particularly problematic, but the results were still disappointing. He had also used the space under the piano as a storage area for unused boxed equipment. (We got the impression that the boxes were primarily there for convenient storage rather than acoustic treatment!)



Clive's difficulties in recording a usable solo piano sound were compounded by the fact that his aging Yamaha Promix 01 digital mixer (above) was rather the worse for wear. The preamps and outputs were much too noisy for this critical recording task, so a much quieter Focusrite Voicemaster Pro preamp (right) was brought in to eliminate this problem from the proceedings.

Hugh and I arrived on schedule at Clive's London home and, realising that the chocolate-biscuit budget had all gone on the piano, we set straight to work fortified by an excellent mug of coffee. The most logical first step was to get Clive to recreate his normal recording setup, which was based around a coincident pair of AKG C414ULS mics set in cardioid mode and fed into a Yamaha Promix 01 digital mixer, and then into an Aiwa portable DAT recorder via its S/PDIF input. The piano lid was fully open and he had tried a number of mic positions, varying from having the mics actually inside the piano to setting up around a metre away. He'd also tried close-miking the strings with the mics in omni mode, but had observed that the problem with close miking was that the mechanical hammer action and pedal noises were too dominant.

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Finding The Weakest Link

Hugh and I set out to make some test recordings using Clive's original setup and discovered that the recordings did indeed sound strident and lacking in depth. Certain upper mid-range notes were particularly prominent and the piano didn't sound tonally balanced at all — you could hear these effects in the room as well, but the miking accentuated these faults dramatically.



A significant proportion of any recorded piano sound is made up of sound reflected from the soundboard to the listener off the floor, so Clive's decision to stack all his empty equipment boxes under the piano was making a serious impact on the sound. The reflections from the side wall were also causing coloration, so both problems were addressed by moving the piano a little away from the wall and then stacking the boxes between the piano and the wall to reduce reflections.

Not only that, but the Promix 01, which had been secondhand when Clive bought it, was incredibly noisy for some reason. Not only were the mic amps hissy and hard-sounding, but there was also a considerable amount of noise on the stereo input channel connected to the DAT machine for replays. These problems were made all the more obvious because of Clive's nice PMC AB2 monitors, driven by a huge Bryston 4B amplifier. When it was launched, the Promix 01 was an excellent budget mixer, but I think it's fair to say that the mic amps were not brilliant and certainly not what you'd choose to do justice to a grand piano that costs around the price of a decent flat! Furthermore, the Promix 01 has only 16-bit converters, so getting adequate resolution on an instrument with such a huge dynamic range is a bit of a struggle. Add this to the deterioration that several years of hard use had wrought and it was little wonder that the sound quality was less than optimum.

As luck would have it, we had a Focusrite Platinum Voicemaster Pro recording channel in the car, and Hugh had also brought along his collection of Sennheiser MKH mics plus a couple of BLUE Baby Bottles. These and some tall and solid

K&H mic stands were brought into the studio to do some tests, with the piano still set up as Clive had left it. Because we only had one preamp, we had to conduct our tests in mono, but working on the KISS principle (Keep It Simple Stupid!) actually helped focus the work on getting a good raw sound, and once we had that, adding a second mic for stereo would have been relatively easy.

The first mic we tried was a Blue Baby Bottle (and we also tried a pair of these via the Promix 01), but because this has a 2dB frequency response bulge in the 1-3kHz region it tended to exaggerate the stridency of certain notes, which wasn't helping. So we decided to switch to a mic which we both know well and which has an absolutely ruler-flat response — the Sennheiser MKH40 cardioid mic. Checking this with both the Promix 01 and the Platinum preamp confirmed that the Promix 01 mic preamps/converters were not doing anyone any favours, and straightaway we got a more accurate impression of the sound though the Focusrite, but this clearer recording also affirmed that we had some room acoustic problems and mic placement issues to deal with. We also tried one of Clive's AKG C414s, again in cardioid mode, through the Platinum preamp. This eliminated the noise and muddiness that had previously plagued Clive's recordings, though the slightly forward character of the mic gave the piano a somewhat hard, thin tonality that no amount of juggling with mic positions and distances was able to resolve with the current room setup.



Reducing the degree of reflection in the comparatively small room was an important concern, in order that a hall reverb could be added at a later stage, so duvets were hung in a selection of places, including behind the pianist in the corner of the room.

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Evaluating The Recordings

The noise problems with Clive's Promix 01 console made it very hard to evaluate the (mono) test recordings we had been making, so we retired to Clive's lounge where he has a rather good hi-fi system set up, again based around PMC monitors — this time the floor-standing FB1s driven from a Bryston 4B amplifier. We were also able to compare our test recordings with some of Clive's favourite commercial piano recordings, where it was immediately evident that the room acoustic of a concert hall played no small part in the overall sound. Being realistic, we were never going to achieve exactly this sound in such a small and acoustically difficult room, but we felt that if we could tame the worst of the room problems and develop a suitable miking strategy, it might be possible to get close to the desired sound (Clive wanted to record both jazz and classical pieces) by adding a small amount of concert-hall reverb from a good-quality digital reverb processor.

Clive revealed that his plans for the future involved recording everything directly into his computer via a good-quality soundcard (24-bit resolution would be worthwhile with piano). He had already purchased some Cat-5 extender boxes for his keyboard, mouse, and monitor, allowing the CPU to be located in the next room to keep out the fan and hard-drive noise. His hope was that he could use a reverb plug-in to produce the necessary concert hall ambience, but I told him that, in my experience, none of the native plug-ins other than the most power hungry were as good as a decent mid-priced hardware reverb unit, simply because of the CPU power available. This situation improves with every generation of computer, but given the current state of play and Clive's budget, my own view was that he should either buy a mid-priced Lexicon or TC Electronic unit or go for plug-ins that run on a DSP card, such as the TC Powercore, which I use myself and know is up to the task reverb-wise.



One of Clive's own AKG C414ULS mics was used in the first instance for experimenting with positioning and room treatment, but its frequency response tailoring made such critical evaluations difficult.

Both Hugh and I also stressed that he would have to buy some new preamps, but given that the room would never be perfect and would probably always be the limiting factor in the recordings, we saw little point in buying anything too esoteric. As it was now clear that the Promix 01 had to take early retirement, a good-quality, small analogue mixer seemed to be the most practical solution, as this would handle Clive's modest monitoring and mixing requirements as well as providing a few adequately good preamps. Hugh suggested one of the smaller Mackie VLZ Pro models, as he uses one himself and finds the preamps to be extremely good. If a mixer hadn't been required, then preamps of the quality of the Focusrite Platinum series we'd brought along would have been ideal, though the EQ and compressor sections would have been somewhat redundant in Clive's case, as he planned to record flat and then do any necessary processing via software.

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Back To The Room

Having auditioned our various test recordings, Hugh and I had a much better idea of where the problems lay. We felt that the first thing we had to do was sort out the basic sound from the piano in the room. In a concert situation, a grand piano typically expects to 'see' a hard reflective surface (usually wood) below it, plenty of space around it and also a high ceiling. In fact, a lot of a piano's sound comes from the bottom of the soundboard, reflecting off the floor, but with all the boxes stacked underneath Clive's piano that contribution was being soaked up here. So, our first job was to remove the boxes, and immediately we heard a very significant improvement to the sound. The instrument sounded better balanced,

and much of the hard peakiness had gone. We also felt that the side of the piano being hard up against one of the side walls may not have been a good thing, so to make the side wall 'go away', we moved the piano about eighteen inches into the room, and used all those cardboard boxes to build an absorbing wall between the piano and the actual wall of the room.

The corner behind Clive's piano stool was deadened somewhat by hanging up a duvet and Clive had also rigged up a rail so that we could hang a double duvet close to the wall facing the open piano lid. A further duvet was draped over the rack holding the computer monitor, close to the tail end of the piano. The acoustic sound of the piano in the room was clearly better now, so we set about making some more test recordings.



A Sennheiser MKH40 microphone was then selected for further positioning and room-treatment tests, because of its ruler-flat frequency response.

We reverted to Clive's AKG mic now that we knew we were on the right track acoustically, as this was what Clive would be using. The C414 is widely used for piano recordings anyway, and it should be able to deliver a good sound if you find the best position. We started with the AKG mic around one metre from the open lid and aimed at a point around halfway up the lid. While Clive played, Hugh moved the mic around, while listening via Clive's very old Pioneer headphones — which were less than accurate, so we added some decent phones to his shopping list! Hugh recommended Sony's MDR7506 as a good benchmark, but suggested that Clive try out several other manufacturer's offerings in comparison to find something that he felt comfortable with, both sonically and physically.

A number of miking options were tried with the mic closer to the piano, around towards the tail of the piano, and also with the lid half open as opposed to fully open. When we took these test recordings into the lounge and compared them with our earlier efforts, the piano sound was definitely better, and the strident peakiness that plagued the earlier recordings was much less pronounced. However, we still felt the overall sound was a little lightweight and some notes still sounded better than others.

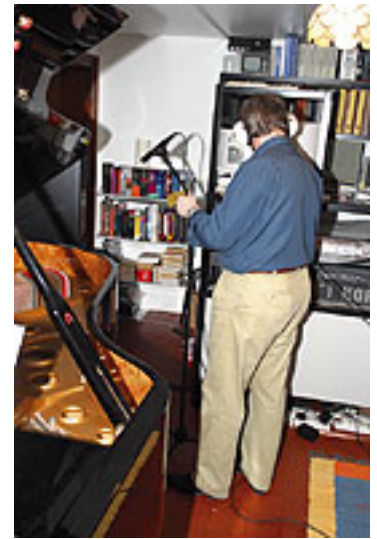
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Using An Omni Pattern

At this point I felt it would be worth trying the mic in omni mode, even if it meant putting some absorbent material behind the mic to reduce the effect of the room. When recording acoustic instruments, omni mics invariably give a more open, honest sound than cardioids, the only downside being the additional amount of spill or room noise picked up. Hugh had said earlier that in a high concert hall he

would often set the mics up a couple of metres or more from the piano and a couple of metres from the floor, so trying to work this technique into the present situation, I felt it would be worth trying something like this with the mic set to omni mode, but with a pillow positioned between the mic and the ceiling to attenuate immediate ceiling reflections. Furthermore, I reasoned that just as in the case of a boundary mic, if we could get the capsule fairly close to the ceiling, we would reduce the possibility of ceiling reflections getting back into the mic from anywhere other than directly behind the mic.

The setup we decided on was with the mic about 2m from the piano, facing the middle of the fully open lid and around 25cm from the ceiling. The hanging duvet was directly behind the mic, and I held a pillow above the mic while we made our test recordings. As soon as we heard the playback, we knew we were onto something very promising with this technique. During the previous tests, Clive had commented that he hardly recognised the recorded piano sound as his own Fazioli, but now the whole thing sounded more integrated, more tonally even, and there was a sense of the performance taking place in a real space. We still didn't have the concert hall — that would have to be simulated using electronic reverb — but we were at last getting a decent basic piano sound.



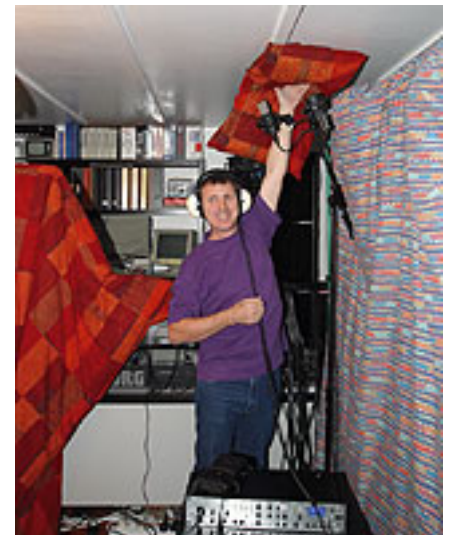
A number of different mic positionings were tried whilst monitoring via headphones, including miking from the tail of the piano.

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Recommendations

Holding a pillow against the ceiling during every recording is plainly impractical, so we suggested an area of thick acoustic foam above the position where the mics would be set up, with a few additional squares of foam placed in a chequer-board pattern above the piano to reduce the ceiling reflections to a more manageable level. Even though the mics wouldn't pick up much in the way of ceiling reflections directly, these would still colour the overall acoustic of the room.

All our tests had been in mono, and Clive obviously wanted to make stereo recordings, so Hugh suggested using the AKGs as a spaced pair, still in omni mode, with a fairly small distance of separation. Finding the



Once the main room problems had been tackled, Hugh and Paul switched back to using Clive's AKG C414ULS for further positioning

correct spacing would involve a fair amount of trial and error, but he felt that the range to try would be starting at around 30cm and working up to a maximum of a metre. A spaced pair is less mono compatible than a coincident pair, which is how Clive had been recording originally, but it tends to give a richer stereo image and a better sense of space. In any case, given that omni mics seemed to give the best sound, the coincident option was no longer viable, as this technique requires directional polar patterns.

tests. In the end, switching the mic to an omnidirectional polar pattern and placing it near the ceiling provided the best sound, although a pillow needed to be held in place behind it by Paul to stop ceiling reflections adversely affecting the sound.

The curtain pole holding the duvet over the wall opposite the piano lid would be made permanent, with the idea of hanging two double duvets side by side on sliding curtain rings, and Clive said he would also like to try a further thick duvet over the wall adjacent to the piano to see if he could get away without having to keep the wall of cardboard boxes. If it didn't, the boxes would stay, though he would probably hang a curtain over them to tidy up the appearance of the room. In any event, the boxes would not go back under the piano, as they'd clearly compromised the sound. A thicker and larger duvet would also be hung across the corner nearest the piano stool, and all the duvets would use light-coloured covers to maximise the available light in the dark basement room. Bass trapping was deemed unnecessary, as the room has one window and three fairly lightweight doors leading into it, all of which allow low-frequency energy to pass through.

On the equipment front, the priority was to pension off the Promix 01, add a small but good-quality analogue mixer (for its preamps and monitoring controls) and also look at good-quality reverbs, possibly DSP card-based. **SOS**

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Clive's Comments

"Paul and Hugh quickly focussed on several problems, including the room acoustics and the Promix 01's pre-amps. While I'd come to realise that those two areas probably needed attention, I couldn't tell how to proceed overall because there were too many things I didn't know. Firstly, I didn't even know what would be a realistic goal — a key question was whether, in that room, professional-grade recording equipment could make recordings that wouldn't sound out of place amongst recent jazz or classical solo-piano CDs. Sadly, Paul and Hugh tell me the answer is 'No' — the room will always be too limiting, no matter what acoustic treatment is applied, so it would be pointless using a wonderful £1500 stereo preamp, because the room would make it sound hardly better than one costing £500. It would seem that a good budget solution, such as a Mackie VLZ Pro or Focusrite



Platinum, is all that my recording environment warrants.

"Secondly, there were the technicalities of recording to deal with. I didn't know whether I was failing to deal properly with the room acoustics and mic positioning, or whether my mics were unsuitable. I learnt that my C414s used in omni mode and effectively positioned, in conjunction with some uncomplicated acoustic treatment of the room, could produce a much better sound than I'd come to hope for. However, I think Hugh's Sennheiser MKH40 sounded a lot better!

"While the severe limitations inherent in the room are disappointing, I do now know I can greatly improve my recordings without spending a lot on equipment. On the other hand, having heard how much truer the recordings were with Hugh's MKH40, I also know that better mics than mine could add further improvement, in spite of the room and still using only good-quality budget preamps. I still have to decide which sound card to get, but thanks to Paul and Hugh's visit, my attention is now far more narrowly focussed and I can hope to make good headway. This session really was very valuable to me — every home studio should have one!"

Published in SOS December 2003

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STUDIO SOS

Steve Graham

Published in SOS August 2002

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People : Studio SOS

The SOS team answer a reader's cry for help, and assist him to improve his recordings of guitar and vocals.

Paul White

Following the lead of the numerous home and garden make-over programmes that seem to be taking over the television schedules, when SOS reader Steve Graham sent us an email asking for advice on some recording problems he'd been having, SOS Technical Editor Hugh Robjohns and I decided to pay him a visit at home, to see if we could come up with a solution on the spot. Armed with a couple of preamps, an assortment of mics and a bag full of cables, we set off for Bristol where Steve had his studio set up in a small bedroom.



Following a previous enquiry to SOS, Steve had already equipped himself with a pair of CAD M179 multi-pattern microphones, which he was using via the mic preamps of his Mackie 1202VLZpro mixer, into his Roland VS1680. He'd already decided that the Roland mic preamps weren't sensitive enough and he felt the quality of the Mackie preamps was better, but he still wasn't getting the result he wanted. Unfortunately, one of his newly acquired mics had mysteriously stopped working, but, as luck would have it, Hugh had brought along one of the same type (but in a fetching shade of blue), so we were able to continue.

The Problem: Recording Guitar & Vocals Together

On the face of it, the problem seemed simple. For this particular project, Steve wanted to record just his acoustic guitar and vocals together as a live performance, but this was complicated by two factors. Firstly, he tends to play with the guitar fairly high up his body, so there's little physical separation

between the vocals and the guitar. Not only does this make balancing difficult, it also opens up the possibility of phase problems, because the vocal mic picks up a significant amount of guitar.

In this kind of situation, the ideal is to space the mics apart by at least five times the distance between mic and source, but here the spacing and mic distance ended up being about the same. The second complication was Steve's playing style, which features some very nice finger picking punctuated by rhythmic damping as he slaps the heel of his hand onto the strings near the bridge. While this sounded fine live, when you came to record it the guitar sound was dominated by loud, deep thumps that peaked 12-15dB higher than the average guitar level.

Steve had tried recording in his usual standing position and, after reading up on recording techniques in SOS, he'd constructed an absorbent corner in the room using two up-ended mattresses. This certainly helped with room reflections, but the close proximity of the mics and the heavy string slapping were still giving problems. He'd started off using the mics in wide cardioid mode and then done a few experiments with figure-of-eight mode, but the improvement wasn't nearly enough. It was at this stage that we decided to lend a hand.



Experimenting With Mic Positioning

After we'd arrived, introduced ourselves and downed the obligatory coffee (Steve also came up trumps with chocolate biscuits, just in case anyone else is thinking of asking us round!), Steve played a couple of songs through to us as though performing live so we could see what we were dealing with. We immediately identified the high position of the guitar as a problem, so we asked him to try playing and singing on a tall stool where the guitar could be positioned a little lower without unduly compromising his playing style. He also tended to sway and move quite a bit when playing standing, which meant close-miking was a problem, so sitting on the stool also helped here, by reducing the amount of physical movement, allowing us to place the mics a little closer for better separation without fear of him swaying out of range.

We patched his mics, still in cardioid mode at this time, into the two SPL Channel One preamps I had brought with me, set with no compression or EQ at first so as to approximate what he'd get if he used the Mackie preamps. The reason we didn't use his Mackie preamps straight off is that I wanted to have the SPL's EQ and compression at my disposal if we needed it.

The guitar mic was set up around eight inches away from the instrument, aimed roughly at where the neck meets the body, while the vocal mic was slightly closer and equipped with a pop shield (laddered, but still effective). The lowering of the guitar, combined with moving the vocal mic to just

above mouth level, increased the separation to the point where it was possible to get a good guitar/vocal balance, and the general quality of the recording was good, with no obvious room coloration.

However, that slapping action was still clearly causing problems. At this early stage, we were making sample recordings both to Steve's Tascam DA20 DAT machine and to the VS1680 (to eliminate the VS1680 from our enquiries!) and, though the DAT seemed marginally clearer, the difference was not significant enough to worry us. What did confuse at first though was that the headphone output of the Roland VS1680 sounded noticeably less clear than the headphone output from either the Mackie mixer or the DAT machine, though this turned out to be a problem only with the headphone amp and not with the general quality of the recording. Steve was using 80(omega) Beyerdynamic DT250 headphones, and it is possible that the fairly low impedance may not have suited the Roland too well. To solve the problem, we took the Roland VS1680's main outputs through the Mackie's tape return and used its headphone amp to check our playbacks.

Dynamics & EQ Settings

While Hugh checked the mic positions, I set up the dynamics and EQ sections on my SPL preamps to do



Test recordings made to DAT while setting up revealed problems which needed flexible equalisation and dynamics to sort out, so a couple of Paul White's external preamps that had been brought along were pressed into service to help out.

something about that thump, so I ended up with around 8dB of cut at 120Hz on the guitar mic plus a couple of decibels of the SPL's Air Band EQ boost (a wide-band 14kHz peaking filter). Then I adjusted the compressor with Steve playing, so that only a nominal amount of gain reduction was showing when he played normally, but the compressor really slammed in when he started slapping. The SPL compressor is an automatic soft-knee design, so I only had to adjust the amount of compression and



Here you can see where cardioid microphones were found to give the best results — note the up-ended mattresses in the background, being used for some rough-and-ready acoustic treatment.

the make-up gain. From the gain reduction meters, I'd say there was between 10 and 12dB of gain reduction applied during those slaps, but the recovery was so fast that the rest of the guitar playing was unaffected.

The vocal was treated to the barest hint of compression and a decibel or so of Air Band EQ, and the 75Hz low-cut filters were switched in for both preamps (the high-pass filters on the mics were switched out to allow experimentation from the preamps, although we would have had the same result had we used the mic's filters).

Straightaway, the result was far more to Steve's liking, with the thump tamed and the Air EQ adding clarity without harshness. The final step was to pan the vocal track just left of centre and the guitar track to around three o'clock, which produced a reasonably well balanced sound with a hint of stereo spread. No effects were added at this stage. In isolation, the guitar track sounded a little bass light, but when the spill from the vocal mic was added in, the tonal balance was actually rather good.

Decreasing Spill Using Figure-Of-Eight Patterns

Though we were getting very usable results by this time, both Hugh and I were keen to try using the mics in figure-of-eight mode, because this way it's possible to exploit the off-axis null of the figure-of-eight pattern to improve isolation. To do this, we set up the vocal mic so that it's 'deaf' axis was pointing directly at the guitar body and the guitar mic was set up so that its deaf axis was directly in line with Steve's mouth. Of course a figure-of-eight mic picks up equally from both sides, so to avoid problems from room reflections, we hung a duvet from a mic stand, directly behind the mics. This meant Steve was almost totally surrounded by acoustic treatment of one kind or another, hence the photo looking down on him from above!



An improved setup using the two microphones in figure-of-eight mode.

Checking out this setup showed we had reduced the amount of voice spilling onto the guitar mic by at least 6dB, and lowered the guitar spill to the vocal mic by about 3-4dB. While this may not sound like a lot, it reduced the levels of phasing when the two mics were combined in mono, and it also made getting a good balance much easier. The reason the guitar spill was reduced a smaller amount than the vocal spill was due in part to reflections from the ceiling, but also to the fact that a guitar is not a point source of sound. The angle of rejection of a figure-of-eight mic is actually very narrow — rather narrower than the radiating surface of an acoustic guitar body — so unless you're recording sound sources that only radiate along a narrow line, in near anechoic conditions, you won't get anywhere close to the theoretical rejection. It is worth paying close attention to the alignment of the vocal mic to optimise the relative angle of the figure-of-eight null and the axis of the guitar body as a few degrees

difference can increase or reduce the separation by several decibels. The improvement we achieved here changed the balance, so a little adjustment to the pan controls was required to get the vocals back into the centre of the mix.

As an experiment, we repeated the recording substituting a Sennheiser MKH30 small-diaphragm mic on the vocals to see if it would give even better separation. As it turned out, it did improve the separation slightly, but Steve felt it didn't suit his voice as well as the larger diaphragm CAD M179.

At this stage we had a recording that stood comparison with what you'd expect to hear on a record — it only needed a little additional ambience to complete the picture. Before setting up a reverb however, we decided to try the same recording technique using Steve's own equipment instead of the SPL preamps. Although the Mackie mixer gave us a beautifully clean recording, its EQ wasn't flexible enough to sort out the slapping problem (and neither was that of the VS1680), so we advised Steve to put a good-quality hardware equaliser on his shopping list. Steve already owns a Drawmer DL241 compressor, which should allow him to achieve very similar results on the guitar to those we obtained with the SPLs, without having to buy anything more than an equaliser. As it happened, he'd already half decided to buy a two-channel TLA equaliser because of its hybrid tube circuitry and because it included good mic preamps, so we saw no need to advise him otherwise.

Final Touches

Turning now to the 'polishing' part of the job, Steve had a Quadraverb and the effects processing in the VS1680 available, although we soon discovered that Steve was a little unsure of what some of the reverb parameters actually did. We found the Quadraverb was a little too coarse for this Job, but the Roland effects seemed adequate. After explaining the role of the adjustable parameters, I had a go at adapting a Small Room program in the VS1680 to give a more intimate vocal sound that had space without sounding washy. Steve didn't want a 'steamy' sounding modern reverb anyway, as the sound he was after was closer to that of folk records made in the '70s, when most reverb would have come from a plate or from a natural acoustic.

Because there seemed to be no dedicated ambience programs, I thought the best approach would be to crank up the early reflections level to near maximum and reduce the decay density and diffusion to emphasise the early reflections. What we finally arrived at was based on a Vocal Room program with a 1.8s decay time, 80ms pre-delay, medium density (44) and medium diffusion (46) with the early reflection level up at 96. A high cut at 10kHz coupled with 3dB of HF damping (14kHz) rounded out the high end, and by adjusting the reverb level carefully, we were able to get a good sense of space



Hugh Robjohns auditions recordings through Steve's Mackie 1202 VLZpro mixer — although Steve's VS1680 recorder has a headphone output, better results were achieved by monitoring the main outputs through the external mixer.

without the reverb being too obvious. The Room Size parameter was set to 30m.

Steve was pretty happy with the sound we had achieved together, but happened to comment that his mixes never sounded as loud as those on commercial CDs, so a whole discussion of mastering and, in particular, the use of dynamics followed. Though the VS1680 includes some mastering programs, Steve wasn't confident enough to adjust the parameters, so he'd experimented with some likely looking presets, but found the results a little too coloured sounding. Also, as these mastering plug-ins can't be used at the same time as reverb, and as we couldn't take the time to do the mixes, we decided to try a regular full-band compressor to see if this could be used to fill out the sound and push the level.

This was used as a master insert effect and set up with 1.5:1 ratio (the ratio lowest available) and around -20dB threshold. Attack time value was 24 with release time set to 50. The recording was made leaving around 10 to 12dB of headroom, so I guess the effective threshold setting relative to the peak level was somewhere between -5dB and -10dB. As it turned out, this was more successful than expected, pushing up the level and helping to knit the guitar and voice sounds together.



Once the sound had been recorded as well as possible, Paul White helped Steve to create suitable reverb and compression treatments for the mix within the Roland VS1680 multitracker.

As Steve expressed an interest in having his material professionally mastered, we didn't go any further down this road, but I did offer to master a few tracks for him at my own studio and to explain the processes that could be used. He was particularly interested in the possibility of using valve or valve simulation technology to produce more of a retro sound, closer that of his favourite '70s recordings. **SOS**

A Satisfied Customer: Steve's Thoughts On The Session

"The session was a very positive and beneficial experience for me. I think that I'd been heading pretty much in the right direction, but having the SOS team's experience and knowledge on hand really brought about significant improvements. I think the key differences were using the figure-of-eight mics (I'd just bought mine and hadn't really used them until then), the EQ from the SPL preamps, and the tweaking of the VS1680 reverb settings to create a patch that seemed to really suit my voice.

"Also, the conversation we had about me not being able to accurately emulate early '70s acoustic sounds (my reference era!) with my own digital multitracking gear was very reassuring, and I look forward to hearing what the mastering stage can add in helping achieve this. I originally felt sure that we wouldn't get good results using the VS1680, but I was impressed with what we eventually achieved. However, I still think that the sound we got recording directly to the DA20 two-track DAT machine was slightly better. Also I think that having subsequently tried

to emulate the session on my own using my own equipment, the SPL preamps had a fuller, 'meatier' sound than those in my Mackie 1202VLZpro desk, possibly because of their valve circuitry and their integral EQ and dynamics. At the moment I'm thinking that I could keep on using the VS1680 and invest in some better mic preamps and possibly a good reverb unit, but I'll eventually get a stand-alone hard disk recorder and outboard effects.

"Having failed up until now to get a guitar sound I liked (both on my own and in a few commercial studios), I'm now convinced that I can do a good job of the basic tracking at home, even if I end up going elsewhere for mastering (and perhaps even mixing) because of the limitations of my studio space."

Postscript: Mastering The Recordings

A couple of weeks after our visit, Steve sent me a CD-R containing the recordings we'd done together, plus a couple more he'd done after we left using his own equipment, but employing our working methods. I was very pleased with what he'd achieved, so I set about processing all the mixes via my Drawmer DC2476 mastering processor to see if Steve would like the result. Mostly the processing comprised some very gentle overall multi-band compression, using a threshold setting of around -30dB and a ratio of 1.1:1 in each band. A tiny amount of 'air' EQ was added (around 1dB of boost at 14kHz with a Q factor of two) and the three-band tube emulation was turned up to just below half way on each band to try to capture more of an analogue feel. Finally, the compressor output level was turned up until the highest peaks just tickled the output limiter. From my perspective, the result of this processing was a noticeable increase in level, a more even and better integrated overall sound, and a subtle but beneficial enhancement to both warmth and detail.

Steve called back to say he was well pleased with the results on the whole, as the tracks were louder, more transparent sounding, and the bass end seemed better controlled. However, he still felt the sound was slightly too upfront and modern for his taste, (based largely on '70s recordings) which I attributed to close miking in a dead room and the use of non-esoteric reverb. I suggested that a better reverb unit with a good room/ambience program might help create the illusion that the recording was made in a larger space and recorded at some distance from the performer.

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STUDIO SOS

Tim Way

Published in SOS September 2002

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People : Studio SOS

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Studio SOS

Another reader's studio gets the benefit of expert SOS staff attention. This month, it's the turn of Tim Way, whose mixes sound fine in his own studio, but don't travel well.

Paul White

Dorset-based Tim Way got in touch with SOS recently to ask for help with the sound of mixes produced in his studio. The problem was that they lacked sparkle and seemed slightly woolly at the bass end when played back over other systems, even though they seemed OK in his studio. Technical Editor Hugh Robjohns and I loaded up my car with a pair of Mackie HR624 monitors I'd had in for review, some test equipment, and a selection of recordings that we felt we knew the sound of, and set off down to Tim's place to sort out his monitoring problems.

The studio, which Tim uses with his songwriting partner Steve Shepherd, is located in a converted



bedroom, where a significant part of the room has been walled off to create a large vocal/instrument booth. The resulting L-shaped space is a little cramped, and after initially setting up the room working lengthwise (the speakers on one of

the narrow walls), Tim decided to move things around. The main impetus was that he felt the stereo image to be too narrow and uninspiring when he was composing and mixing.

Turning the room around allowed the monitors to be set up either side of a large keyboard/outboard stand, more or less in the corners of the long wall. This worked out better than expected, despite the proximity of the vocal booth's glass window directly behind the mixing position, possibly because the speakers had to be spaced further apart than ideal. Tim decided to mount the speakers horizontally on their stands, and because they were so far apart, he had to angle them in more steeply than usual. This, combined with the narrower treble dispersion when speakers are placed horizontally, seems to have resulted in the glass window causing less of a reflection problem than expected. Unfortunately, he'd also used foam acoustic tiles covering most of the original end wall, so now these were on his left with a more-or-less bare reflective wall to his right -- not good for symmetry.



Tim Way in his home studio. The acoustic tiles you can see behind him were left from a previous studio setup, where the sitting position had been facing that wall. However, with the newer equipment configuration these tiles were causing an imbalance in the stereo image.

Though this didn't seem to be causing any obvious problems, an impromptu experiment with a duvet and two wooden battens to deaden the live wall indicated that the stereo imaging would be improved a little if half of the original tiles were removed from the left-hand wall and relocated to the right, to restore some acoustic symmetry. As the glass control room window wasn't causing any serious problems, we sugg

ested that Tim might think about fitting a wooden slatted blind which could be left half open when mixing, to help diffuse any reflections.

The first main thing to do was listen to some of Tim's mixes and check these against some of our test recordings. Our reference material quickly revealed that in this particular room, and with Tim's choice of hi-fi amp, the Spirit Absolute passive monitors were producing very little true bass (despite being in corner positions) and they also sounded quite aggressive in the upper mid-range. Because of this, Tim was tending to mix with more bass than necessary to compensate and he was also backing off on the bright sounds, which is why his mixes sounded dull when played

elsewhere. Furthermore, because the monitoring wasn't adequately accurate, Tim was using quite a lot of EQ to try to get sounds to fit, and in some cases this turned out to be counter-productive.

As an experiment, we set up the Mackie HR624 active monitors we'd brought with us -- I have HR824s in my own studio, so I'm familiar with the general character of these speakers. We expected the sound to be different in some respects, but we were surprised at exactly how much difference there was. Suddenly there seemed to be about another octave of bass, and all the upper harshness disappeared. Considering the unplanned acoustics of the room and its currently asymmetrical setup, what we were getting back from the speakers sounded surprisingly accurate on our known recordings. That isn't to direct too much blame at the Spirits, because the performance of any passive monitor can vary enormously depending on the type and rating of amplifier used to drive it. The Mackie speakers also cost considerably more than the Spirits, and monitoring is generally a case of 'you get what you pay for'.

My own feeling is that Tim's hi-fi amp wasn't really powerful enough for the job, in which case a lack of bass and high-frequency roughness is what you'd expect. At a later stage we tried the Spirits th



Because of the wide equipment rack at which Tim worked, his speaker stands were spaced further apart than would be ideal.

e right way up and found the sweet spot widened significantly (as expected, because of the greater horizontal dispersion), but the tonal problems we'd experienced originally were still there.

Into The Mix

For the rest of the session, we reverted to the Mackie HR624s. Now that we had a monitoring system we could trust to be reasonably accurate in Tim's room, we set about analysing a mix he was working on and came up with a few suggestions. The first step was to strip out the EQ on his tracks to see what the source material was like. Even with the new monitoring, the bass end of his mix wasn't working too well, so we concentrated mainly on the kick drum sounds and the bass instruments, in this case an electric bass guitar and a bass synth arpeggio. Everything was recorded into Pro Tools LE, so we were able to use plug-ins for testing out our ideas on EQ and dynamics. Any MIDI instruments not recorded as

The Importance Of Good Monitoring: Tim's Comments

"Because Paul and Hugh sorted out my problems really quickly, I didn't realise how much I got out of the visit until they had gone. In particular, I was really impressed by the huge difference the change of monitors made, so I decided to go out and buy a pair. After a week's use I'm pleased to say that they remain a joy to work with -- I had no idea a decent pair of monitors could make such a difference. All I've ever wanted was to hear what I was mixing, but it seemed such a tall order within the confines of a bedroom studio. Cheers guys, I'm in control and loving it!

On the songwriting front, Steve and myself have just joined the British Academy of Composers & Songwriters. We will be attending regular workshops and are

audio were also driven from Pro Tools LE, which meant we also had the opportunity to experiment with different sounds as necessary.

First off, we felt that the kick drum wasn't working.

It was a sample that Tim liked because its attack had an acoustic quality, but it didn't have much punch and, as the track had a very dance-style rhythm section, he'd tried to add

ress the sonic deficiency with EQ. The problem was that piling on enough bass lift to bolster the LF energy had made the sound quite woolly, so we suggested he might be better off using a combination of two samples -- one with the attack quality he liked, and another with a very deep sound, such as a TR909 kick.

We trawled his keyboards and sample CDs for a suitable kick, but surprisingly few of them were solid enough. In the end, we found one that also included (unwanted) vinyl noise that we felt would work well enough just to prove a point. Layering this with the original kick Tim had chosen produced a much deeper, more contemporary sound, and instead of having to use a lot of EQ, we found we could create a wide range of tonality simply by adjusting the balance of the two samples. I messed around with a gate trying to reduce the amount of vinyl noise following the sample, but this wouldn't have been necessary if a more appropriate sound had been found in the first place.

Moving to the bass instruments, we tackled these one at a time. working on the bass guitar first, as this seemed to be getting lost in the mix once everything was playing. As is so often the case, the audible part of the bass sound was somewhere in the lower mid-range, so peaking up the EQ at 367Hz gave us a sound we could hear as well as feel. Some very deep bass was rolled off, as the DI'd bass track had too much sub-70Hz energy, and then, instead of using normal compression to even up the sound, we used the *Waves L1 Ultramaximizer* plug-in to limit the peaks quite heavily, with between 5 and 10dB of gain reduction showing up. This dramatically improved the audibility and presence of the bass guitar, allowing it to carry the track while still sounding tight and controlled.

When we brought in the bass arpeggio, it conflicted with the bass guitar rather too much, so we applied 18dB of low cut using a *Waves Renaissance EQ* plug-in (high-pass mode, shelving frequency 980Hz, Q 0.71). T

his stripped all the low end out of the sound but still retained the musicality and the rhythmic element of the arpeggio, and these aspects of the part seemed to work nicely alongside the bass guitar and rhythm track.

working towards a publishing contract. Check out our web site at www.studio42.uk.com -- feedback is always welcome, and we're always looking for collaborators."



Following the visit, Tim decided to upgrade his monitoring, and bought himself a pair of Mackie HR624s.

To add a little more sparkle, without cluttering the track, we brought up the level of the hi-hat loop (a rather processed sound from a sample library) and again limited this, using another *L1* plug-in, to firm it up a little. To help clean up the mid-range, we then filtered some low end out of a string pad so that it wouldn't muddle the mid range. Tim had already identified an irritating edge in the string pad, which he'd notched out using EQ, so the low cut was all we really needed to do. The string pad sounded extremely wide, and when Hugh checked for mono compatibility we discovered a great deal of the high sparkle disappeared from the strings in mono. Reducing the stereo width slightly helped to create a more consistent sound, although the effect could not be removed completely, as it was an integral element of the effect processing of the string pad. If you are hoping for radio play it is important to check mono compatibility, as there are still a very large number of mono radios out there!

A little gentle 'air EQ' was applied to the vocal track, which was quite heavily processed already. This entailed using a Waves *Q4* equaliser to add around 1.5dB of presence boost at 3.5kHz and 7dB of 'air' boost at 13kHz. These settings definitely provided better definition and clarity for the vocal part, but we had to take care not to process it so much as to exaggerate the natural sibilance of the voice. The vocal was already compressed, so we didn't need to change that.

Final Touches

After getting a rough mix together, we made a test CD-R and played it in the car stereo, to get a feel for the overall tonal balance, and found that we'd erred slightly on the side of being too generous with the bass. However, bass

level problems are common in small rooms, so this came as no surprise. A little further experimentation with levels redressed the problem, which is why it's essential always to get used to your monitoring system by playing known commercial recordings over it whenever possible.

So, as always, the real problem did not have a single overriding cause, but was due to a combination of smaller factors. Certainly the chosen monitoring system wasn't producing accurate results in that particular room, but the problem was compounded by the fact that Tim often used large amounts of EQ to try to bend sounds into shape, rather than finding sounds that worked properly in the first place. In other respects, the songs were very well written and arranged, and it was all I could do to stop Hugh singing the track we'd been working on during the long drive home! **SOS**



In order to make a more symmetrical stereo image, Hugh and Tim cleared some space to the right of the workstation and hung up a large duvet to reduce reflections from the untreated wall.

Glossary

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Published in SOS September 2002

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STUDIO SOS

Nick Tucker

Published in SOS October 2002

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People : Studio SOS

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Studio SOS

The SOS team rushes to the rescue of a reader in Somerset suffering from boxy vocals, a weedy mix, and a dodgy tweeter...

Paul White

After reading the first of our Studio SOS features, Nick Tucker sent us an email entitled 'I Want One Too!' and suggested that we might be interested in sorting out his vocal recording and mix problems. So Technical Editor Hugh Robjohns and I arranged to visit, taking along a few bits and pieces of studio gear which we thought might help out.

Nick is a Barrister specialising in criminal defence, but has reduced his work load in order to allow himself more time in his studio, which, unusually, is in his girlfriend's house — at least she knows where he is, I guess! The house is in fact a converted school located in a rather picturesque part of the Somerset countryside and Nick's studio is in what used to be a cloakroom. This presents a challenge right away, because, although the room is some 15 feet long, it is only around five feet wide, so Nick works across the room and has all the equipment and



The boxiness of Nick Tucker's vocal recordings was tackled by adjusting the mic positioning and by holding up a duvet behind him to damp down excess room reflections — anything to keep Editor In Chief Paul White out of mischief...

monitors mounted low down so that he can work while sitting on the floor.

When Nick first called us, he said that his vocal recordings had a somewhat boxy quality, despite being recorded using a Neumann TLM103 microphone, and he was also a little unhappy with his mixes insomuch as he felt the sounds didn't gel properly and were also lacking the clarity of commercial mixes. To solve these problems, we needed to look both at the recording chain and the physical way in which the vocals were being recorded.

Mic Positioning

At the heart of the studio was a Tascam 388, which is a kind of giant Portastudio with an open-reel eight-track analogue tape recorder built into the mixer. It uses Dbx Type I noise reduction, and Nick employed a separate Behringer mixer to combine the output from the

388 with that of a drum machine and a selection of effects units, including a rather nice Lexicon PCM90 reverb processor, an Aphex Model 109 parametric equaliser and a couple of TL Audio equalisers.

Nick describes his music as pop with a folk influence, and he both sings and plays guitar. In the case of the acoustic guitar, he was tending to use a mixture of DI and microphone, whereas electric guitars were often DI'd via a Line 6 Pod and bass parts via a Line 6 Bass Pod, which Nick is particularly fond of.



A fair amount of high-frequency information was being lost by Nick's old Tascam 388 — a brief inspection revealed that the heads were quite worn — but there were also problems with the Dbx noise reduction.

For vocal recording, Nick was setting up his microphone at the opposite end of the room to the equipment, in a corner that he had treated with acoustic foam tiles. The TLM103 was kitted out with a pop shield, but no shockmount, and a Joe Meek VC3 preamp/compressor was in the signal path between the mic and the 388 recorder. To allow us to assess the extent of the problem, we had Nick redo the vocal to one of the tracks he'd been working on using his normal working method. Sure enough, it sounded a little boxy, but it didn't take too long to pin down the causes. Firstly, Nick was standing rather further from the mic than was ideal for such a confined recording environment (around two feet), so the room resonances were intruding onto the recorded sound. To compound this problem, two acoustic guitars hanging on the wall were vibrating in sympathy with his voice and adding to the unwanted resonances. This latter situation was quickly fixed by removing the guitars from the room. At the same time, we reduced the mic distance to around nine inches and moved the pop shield to within a couple of inches from the mic. Note that you shouldn't place a pop shield right up against the mic body — they work much better if there is a reasonable gap between the shield and capsule of around a couple of inches.

All About Dbx Noise Reduction

It seems hard to believe now, but at one time the type of noise reduction process you used was as vital to the success of an analogue multitrack recording as the size of your hard drive is today! Professionals enjoyed the relative merits of noise reduction systems such as Dolby A (or the much improved Dolby SR later) or Telcom C4, while the home studio tended to use equipment fitted with Dolby B or Dolby C (and a few late models with Dolby S), or with one of the Dbx formats (Type I or Type II).

The basic idea of all these systems was to reduce the dynamic range of the input signal in some way prior to recording to tape. On replay the inverse process was used to restore the original dynamics (more or less) while also reducing the apparent tape noise into the bargain. The success of such systems was variable, though, and the best (Dolby A initially, and subsequently the remarkable Dolby SR) are technically complex and tackle the audio in several independent frequency bands to maximise control and minimise modulation side-effects.

The Dbx noise reduction system was relatively simple in comparison and, although not so widely used in professional circles, was adopted readily in the semi-pro markets. These Dbx systems, developed by David Blackner in the early 1970s, essentially employ a wide range 2:1 compressor on the record side, with a complementary 2:1 expander on replay. Since the 2:1 ratio is used over the entire dynamic range of the input material, there is no critical threshold requirement, and so no need to accurately match record or replay levels when tapes are moved between machines. This has always been one of the drawbacks of the Dolby systems, and this is the reason why Dolby tone is recorded at the start of multitrack tapes in professional studios.

In any broadband noise-reduction system such as Dbx, dynamic changes caused by a low-frequency source — a bass solo, say — will result in the level of tape noise being modulated, and with no high frequencies in the recorded signal to mask it, high frequency tape noise will become very obvious. To overcome this significant side effect, the Dbx system applied pre-emphasis (high-frequency boost) on the record side and a corresponding de-emphasis on replay. The idea was to reduce the audibility of high-frequency noise generally, and so the pumping artefacts of the companding process were less audible.

On a good day, with everything set up perfectly, the Type I system was capable of providing as much as 30dB of noise reduction — far more than Dolby A — although it was prone to audible side effects. The difference between the Type I and Type II formats comes down to the details of the level sensing circuits used to control the compression and expansion processes. Type I was optimised for high-quality, wide-bandwidth open-reel recorders, whereas Type II was designed for the rather less reliable results obtained from cassette recorders. Consequently, Type II noise reduction was not as powerful, and more prone to mistracking side-effects.

Whereas the Dolby systems only compressed low-level signals which were in danger of being lost in the tape noise, the Dbx system compressed everything regardless of level and, although far simpler and cheaper to make, this was also the system's primary weakness. If the recording medium was entirely linear at all levels, a 2:1 expander would always undo the dynamic changes applied by a 2:1 compressor, so the replayed signal would be identical to the recorded

signal (ignoring the problems of compressor overshoot and the like). However, as we all know, recording tape is anything but linear, especially at higher levels where tape saturation occurs. The saturation effect means that a high-level input signal is replayed at a lower level and with a different frequency response (and more distortion). Since the expander can only work with the signals found on the tape, the original (but saturated) high-level signal ends up being decoded incorrectly. This is known as a 'tracking error' since the dynamic reconstruction of the expander fails to track that applied originally by the compressor. Tracking errors can also be caused by drop-outs, an azimuth error, or if the machine is not equalised correctly for the tape being used, and typically result in a dull and dynamically compressed (or 'choked') sound.

So, whereas the usual advice is to drive analogue tape fairly hard (even when using Dolby noise-reduction systems) to obtain the best noise performance and the often desirable characteristics of tape saturation, try this with a Dbx system and you will be badly disappointed! For this reason, many recorders that use Dbx have switches to disable the noise reduction on some or all channels, so that good old-fashioned tape saturation can still be used creatively if required. It is also a good idea to switch the Dbx off if recording transient-rich instruments such as percussion, since the companding may mistrack on fast transients.

When Dbx noise reduction is being used, it is generally wise to err on the side of caution when setting levels. Keeping peaks below the point of saturation gives the expander the best chance of reconstructing the original signal properly. This usually means 0VU is the absolute maximum level, rather than the average value to aim for, and recording at a slightly lower level than you would without Dbx isn't really a problem as the amount of noise reduction is so great anyway. It is also vital to keep the tape heads clean and demagnetised, and to have the machine serviced regularly and aligned correctly for the type of tape you are using, to ensure that what goes in comes out properly. *Hugh Robjohns*

Gain Management & Signal Processing

I also asked Nick to set the recording levels as he normally would, because I had my suspicions that the Dbx noise reduction had something to do with the problem. As I guessed, he was setting the levels a little on the hot side and, while most a

analogue tape likes to be 'pushed into the red' (to add a little warmth and compression), this has an adverse effect on the way Dbx noise reduction functions, as any nonlinearities in the recording caused by tape saturation are magnified by the rather extreme encoding and decoding system used by Dbx. For more information on this subject, have a look at the 'All About Dbx Noise Reduction' box.

Having corrected the mic distance and recording levels, we tried another take and found the boxiness much improved, though the vocals still lacked the airy presence that Nick was after. We also felt the 'room boom' could be further improved by placing some kind of acoustic absorber across the room, between Nick's vocal position and the part of the room housing the equipment. Once again we improvised using a duvet, and straight away the sound got noticeably cleaner, as almost all the remaining room resonances and 'colour' were isolated from the mic.

Having got this far, we decided to try the recording again using my SPL Channel One mic preamp, simply because it sounds very open and neutral, and also because it has a good EQ section and compressor. The Joe Meek VC3 is actually a very nice piece of kit, but it is built to a price and consequently loses out in the metering department — there is only a single LED to indicate when compression is taking place, and no gain reduction meter. Furthermore, its optical compressor is designed to impart a definite character to the sound, but we wanted to hear what we were getting as honestly as possible, so that we could determine how much effect the tape machine was having on the sound.

The result this time was more natural, but it became clear the tape machine was losing some of the high-end detail. Nick was also reusing old tapes acquired with the machine, and it was unclear what type of tape the machine had been set up f

or, or indeed when it had been last serviced. Although Nick keeps the heads clean, there was evidence of head wear too, which would also tend to diminish the higher frequencies. So, we added 4dB at the high end using Channel One's Air EQ control (which is simply a wide band-pass boost at around 14kHz) in an attempt to compensate and kept the compression fairly gentle with a gain reduction of no more than 5dB on the loudest peaks. At this point we had something coming back off tape that our ears told us was generally similar to the original performance.

Nick still wanted more presence, so we patched in his Aphex 109 and treated the vocal track to 3dB of 7kHz boost with a 1.5-octave bandwidth and also pulled down the 180Hz region by 1.5dB to clarify the low end. This was definitely closer to the contemporary sound Nick was after and, because his vocal performance is dynamically well controlled, no further compression was felt to be necessary. Nick's singing voice was also free of sibilance, so adding to the top end was vice-free! Interestingly, we tried a similar EQ treatment using the TLA units and found that the sound was quite different and in some ways more 'glassy'. Nick rather liked the effect and resolved to experiment further with the different EQs.



The SPL Channel One's Air EQ control was used to compensate for the high-frequency losses incurred because of the Tascam 388's worn heads.



Although EQ'ing the vocal with Nick's Aphex 109 produced very good results, Paul also tried out the TL Audio Ivory equaliser with similar settings, and Nick found the more 'glassy' sound more to his taste.

Working On Acoustic Guitar

Though Nick hadn't specifically asked about the acoustic guitar sound, I was curious as to how he recorded it, and he told us that, for his Takamine, he used a combination of mic and DI with the mic (his TLM103 again) pointing at the junction of the neck and body. Hugh observed that combining mic and DI can sometimes lead to phase problems that colour the sound, so we suggested that Nick try recording with just the microphone. Nick commented that the miked sound was often insufficiently bright (hence the need to add the DI) so we offered to help hi

m find a new mic position that would give him the kind of result he wanted without having to add in any of the DI sound.

The easy way to find the best mic position is to wear headphones and listen to the mic output as you move it around the guitar, so Nick played and I adjusted the mic. Sure enough, the usual neck-meets-body position was slightly lacking definition so I moved the mic further up the neck. One very interesting result was achieved by miking directly over the headstock from around six inches away — very bright and ringy — but the most natural sound we got was miking just a few inches above the third fret.



Moving the recording mic while listening to the results on headphones allowed Paul to find a better guitar sound.

Acoustic guitars on pop records are often EQ'd to remove the low end and they may also be EQ'd to make them sound brighter than they really are, so we patched in the Aphex 109 equaliser. What we came up with was 5dB of shelving cut at 40Hz combined with 5dB of boost at 15kHz, with the bandwidth control set to its widest position. This produced the characteristic zingy, bottom-light sound that sits so well in pop mixes. A little gentle compression (around 6dB) and a hint of reverb completed the picture. Because we'd eliminated the DI component, the resulting sound was more open and natural than before, and that made the guitar sit well without the bottom end clouding other elements of the mix.

Setting Up A Better Reverb

Next we turned our attention to the reverb treatment. Nick had been using a Hall setting, but felt that something dr

ier and brighter might work better. After a little experimentation with the PCM90 presets, we came up with the Club/Air patch and turned the Club part of the sound down slightly using the Adjust knob (which is normally mapped to this parameter). Right away the sound became better focused and more upfront, and in direct comparison with the original vocal track Nick had recorded using his earlier methodology, the improvement was significant.

Nick had been worried that our recommendations would involve him spending a lot of money, but now it seemed as though a more transparent mic preamp, ideally one with the ability to apply EQ boost in the 14kHz region, and some extra acoustic treatment might do the trick. Hugh also suggested that Nick should invest in a shockmount for the TLM103, partly because the standard mic clip was broken

anyway and this lovely (and expensive) mic was in danger of being dropped on the floor! Hugh also suggested that it would be a good idea to put a plastic bag over it when it wasn't being used to prevent dust getting onto the capsule. It is one of the drawbacks of electrostatic mics that they tend to attract dust if left out in the open, so either pack the mic away in the box — which can be a real pain — or protect it with a polythene bag over the top. A dusty diaphragm is also more prone to suffering from increased noise and frying sounds when the humidity rises.

Fixing The Mix

At this point we adjourned to the local pub for a splendid steak and ale pie, after which we made a start on the mix.

Prior to our visit, Nick had sent us a CD-R of some of his 'work in progress' and, aside from the vocal issues already addressed, Hugh and I both felt the kick drum and bass guitar were a little too far back in the mix. The kick drum was also very clicky and seemed to lack body. On quizzing Nick further about his mixing methods, we discovered that he normally works with an SPL Vitalizer Mk II across the mix buss, so we suggested that he bypassed it while he got the mix sounding as good as possible, and only then switch it in with a view to adding a subtle sparkle and to help overcome the limitations of his old Tascam 388.

To beef up the kick drum sound from the drum machine, we added 6dB of boost at 80Hz combined with around 4dB of cut between 250 and 300Hz — the latter prevented the bass boost from spilling over into the mid-range. The bass guitar needed more definition, so we applied exactly the same strategy as we did on our last Studio SOS and added a little boost at 350Hz to give it more of a 'miked amp' sound. A little judicious rebalancing and the track (which was essentially a guitar/bass/drums mix) started to gel nicely. The drum machine outputs were split into kick, snare and stereo kit, so we were able to add some plate reverb to the snare and kit using a Microverb. We used the Microverb's own EQ to add a little high-frequency boost, so there was some sizzle to the reverb sound. The kick drum was left almost dry — we added just enough reverb to get it to sit nicely with the rest of the kit. The other components of the mix were treated using the same PCM90 reverb patch that we used on the vocal track

Once the mix was sounding good, we returned to the Vitalizer and chose some more gentle settings than Nick had been using previously, with a view to enhancing the overall clarity and separation of the mix. This was a Mk II model that has a few more controls than the regular stereo model and, as I already own and use a Vitalizer, I made the necessary adjustments while Hugh and Nick listened. Firstly, I moved the Mid-Hi Tune control up from its 3.5kHz position to around 7kHz, so as to confine



By using the separate outputs of his Roland R70 drum machine, Nick was able to EQ some extra warmth into the kick sound with his Aphex 109 compressor.

the enhancement effect

cts to the very top of the audio spectrum and to avoid adding harshness to the upper mid-range. The high EQ was set to 10kHz and the boost control set at around half way, while the bass end



After working on the vocal sound with Paul's voice channel, Nick has decided to invest in a similar dedicated unit himself — probably a Focusrite Voicemaster in his case.

was treated to a little of the 'tight' bass boost. Vitalizer users will know that the bass end is addressed via a single control that offers a deep, warm bass when turned anticlockwise from its centre position and a tighter, more focused bass when turned clockwise. The Process level control sets the overall degree of processing, and this was reduced to a setting just below half-way. Switching the Vitalizer in and out of circuit showed that what we were getting now was a more subtle effect that retained the tonal feel of the original more than before, but still aided the clarity and separation of the various elements in the mix. It may be that little or no Vitalizer processing would have been necessary had Nick been using a digital recorder, but in the case of the 388 it definitely helped.

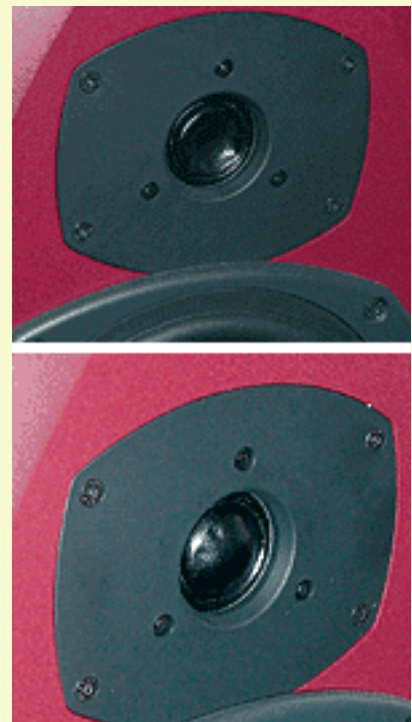
At this point Nick asked what improvements could be achieved using good mastering equipment. We explained t

hat using multi-band compression, top-quality EQ and separate limiting allows the mastering engineer to fine-tune the mix much more effectively than conventional full-band processors. Also, mastering houses invariably have more accurate monitoring systems than project studios, so you get to hear what the recording really sounds like. And of course there's the experience of the guy doing the work and the fact that it is a fresh pair of ears listening to the mix. A good mastering engineer hears what is really there, not what the musician thinks is there! The full details of mastering are too complex to go into here, but we did stress that if he was considering getting any of his mixes professionally mastered, he should leave out any global processing, such as compression or Vitalizer, altogether and let the mastering engineer do the processing his own way.

Homeward Bound

Improvising A Temporary Repair For A Damaged Speaker

For monitoring, Nick had a pair of Tannoy Reveal passive monitors fed from a Samson power amplifier, though he'd managed to push in one of the soft-dome tweeters in getting the speakers from home to the studio. This didn't sound as bad as you might have expected it to, but I thought it was worth trying to fix the tweeter by using Nick's vacuum cleaner to pull the dome back into shape. By gently placing the end of the hose over the tweeter with the cleaner running, the worst of the damage was reversed — but before you try this at home,



What we learned from this visit was once again that problems usually have several causes. The room problems were easily fixed using extra acoustic treatment and by working closer to the microphone, but the use of an aging tape machine with Dbx noise reduction was clearly a major factor in the sound. To get the vocal sound we wanted involved not only sorting out the acoustic problems but also using a more transparent preamp, adjusting the recording levels, using a good quality external equaliser and finding a more sympathetic reverb treatment.

I'm offering no guarantees that this method will work for you!

A brief application of the vacuum cleaner provided a quick fix for a damaged soft-dome tweeter — however, you can still see the creases in the reshaped tweeter, and a replacement would be required in the long term.

Although pulling the tweeter dome back into shape in this way will often appear to repair the damage, any damage caused by the creasing remains and increases the distortion at high frequencies considerably above that of an undamaged tweeter. The best solution by far is to replace the tweeter, and if the speakers are more than a couple of years old or have had a hard life, it's best to change the tweeters in both speakers. For critical work, tweeters should be replaced every five years or so anyway, as they deteriorate over time.

We also discovered that the 388's own mixer section EQ sounded quite harsh and grainy compared to the outboard EQ Nick had bought. Having a Vitalizer turned out to be fortuitous, as it provides a fairly simple way to add 'fairy dust' to final mixes and it also helps compensate for the high-end loss incurred by the tape machine. **SOS**

Session Notes: Nick's Views

"Having been very dissatisfied with the vocal recordings I was making, I have decided to upgrade my mic pre, probably to the Focusrite Voicemaster, and will ensure that, when I record with a mic in future, I rig up some damping akin to the duvet which Paul valiantly held up while recording the test tracks. It made the sound markedly less boxy. It was also helpful to see the EQ settings which Paul and Hugh came up with for both vocals and acoustic guitar on my parametric EQs.

One thing that surprised me was how good my acoustic sounded when miked up about four inches from the third fret. I've always found my Takamine difficult to record, but had never tried this mic placement before. It brought out the top end of the guitar without making it sound harsh, whilst avoiding the boomy sound I was getting placing the mic nearer the body/soundhole of the instrument. Paul and Hugh spent some time trying different reverb presets on my Lexicon PCM90, and settled on one which avoided an obvious reverb tail, whilst lending the sound sufficient ambience to allow it to sit in the mix.

When Paul expressed the view that the Dbx noise reduction on my Tascam 388 was a fairly unsubtle system, and that it might well be having an adverse effect on my recordings, it confirmed my own suspicions, and it has hardened my resolve to acquire a stand-alone hard disk recorder. Hugh also advised me to invest in a balanced patchbay to get the best out of the outboard that I presently wire through an unbalanced patchbay. I see the sense in that and, funds allowing, will try to implement that upgrade at some point.

I have been recording at home for some time without any formal training or guidance, and had become very uncertain as to what I was doing right and wrong. It was extremely useful to have some reassurance that there was nothing I was doing that was obviously daft. I was left with the view that I was heading in the right direction, which was supportive. The tweaks which Paul and Hugh made had a significant impact on the mix, making it seem more balanced and professional.

The opportunity to speak to two people with such experience was invaluable, and has left me with a much clearer idea of what I can expect from the equipment I have, and where its limitations lie. Many thanks to Paul and Hugh — it really was a fantastic experience to have my own personal SOS clinic!"

Glossary

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Published in SOS October 2002

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Studio SOS

Murat Yucel

Published in SOS November 2002

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Studio SOS



A special holiday edition of our hands-on troubleshooting column comes direct from sunny Turkey, where Paul White forsakes the beach to help Murat Yucel refine his band's recording setup.

Paul White

This month's Studio SOS column goes international, coming as it does from the studio of Murat Yucel, which occupies a spare room in his apartment in Dalyan, a small riverside town in southwest Turkey. Imported recording equipment is fairly heavily taxed in Turkey, and the average income seems to be around a quarter that in the UK, so setting up any kind of studio is a bit of a luxury. Furthe

rmore, because of flight luggage weight restrictions, I couldn't take along any rack gear or monitors, or a flightcased Hugh Robjohns for that matter!

The mainstay of Murat's studio is a Fostex VF16 16-track hard disk recorder/mixer. A hi-fi Minidisc machine is used for recording stereo mixes, connected to the VF16 via optical S/PDIF, and there is a small Behringer mixer at Murat's bar (the Blues Bar at the other end of town) which can be brought in when necessary. There's no outboard gear at all, so any processing is done using just the effects in the VF16, and there's not even a loudspeaker monitoring system — everything is done using a single pair of Fostex headphones. However, Murat has some good microphones, including a Rode NT2 with shockmount, an Oktava 219, an Oktava 012 (all capacitor models), a pair of Shure SM58s and an SM57. Other dynamic mics are available

from the bar, where Murat plays most nights with his band, and because the bar operates seven days a week throughout the season and doesn't close until around three in the morning, finding time to record is also difficult. Anything involving 'real' drums must be recorded at the club before opening time, where space and noise is less of a problem than in the apartment.

Murat is a singer/songwriter who plays mainly acoustic guitar, though he can turn his hand to just about anything. Most of the time he records with friends and regular band members, where the instrumentation may be any combination of electric and acoustic guitars, electric bass and hand drum. He likes to record as many parts together as possible, so as to capture a good live performance feel, but in a small apartment room this clearly leads to separation problems. My challenge, therefore, was to try to come up with some suitable recording techniques for the instruments being used and to find a means to record electric and acoustic instruments together without incurring unacceptable amounts of spill.



The Studio Room

I would estimate the studio room is approximately 2.5 x 3m, and 2.7m high. It is built of rendered brickwork as is common in this area and has a tiled concrete floor. At the advice of a friend, Murat had already improvised some soundproofing over the window, compri



sing three-inch thick plastic foam wedged between the window and a wooden board. It was quite effective as a barrier to sound, though the door was less so. This had been fitted with foam draft excluder around top and sides, but there was no sealing on the bottom and the glass panels in the door offered little isolation anyway. As it turned out, the techniques we finally arrived at necessitated recording with the door open, so this ceased to be an issue.

My first modification was to hang a thick rug (these are plentiful in Turkey!) over the window around three inches from the board holding up the foam. Leaving a space increases the effectiveness of the rug as an absorber as both the incident and reflected sound has to pass through it. The idea was that the main vocalist could record with his back to this absorber to help reduce reflected spill and to reduce coloration from the room.

There were a couple of foam cushions in the room already, as well as a block of foam stuck up in one corner on the ceiling. Additionally, there was a rail of clothes hangers with a few shirts on and a shelf full of bottled drinks to provide acoustic (and visual if consumed!) diffusion. These, combined with the rug over the

A kitchen sieve with some nylon stocking over it served Murat as a usable pop shield and, in the absence of enough mic stands, a music stand was pressed into service to hold his Oktava small-diaphragm condenser mic in position.

window and another on the floor, tamed the room surprisingly well, though its low-end characteristics remained somewhat unpredictable. As no low-frequency sounds were being recorded in the room, this wasn't a problem, given that the monitoring was on headphones.

Making Test Recordings

Initially we planned to make a test recording of vocals and guitar recorded together using the two Oktava mics, the 012 on the guitar and the 219 for the vocals. We set up the mics and playing position with Murat's back towards the window, so that the heavy rug would absorb most of the reflections. Murat had been told by a friend with some recording experience to record with the mic very close to a hard surface, presumably to get some boundary-effect low-end boost, but I felt the traditional way would work best, so we kept the mics away from walls but also not in the dead centre of the room so that we wouldn't emphasise any room modes.

There was no pop shield, so we improvised by using a plastic kitchen sieve with a nylon stocking stretched over it. A couple of elastic bands secured this to the mic leaving a couple of inches clearance between the stocking material and the mic — it worked perfectly

with a singer-to-mic distance of around nine inches. However, we did have a couple of practical issues to sort out first. Our first problem was that the capacitor mics refused to work with some new XLR cables Murat had bought. After pulling them apart, it turned out these were wired with unbalanced cable, so properly balanced versions were ordered, along with more mic stands. These arrived a couple of days later, but in the meantime we did a few tests using what cables I'd brought with me. Initially, we had the Oktava 012 taped to a music stand, as there was only one proper mic stand in the apartment (which we needed for the vocal mic), but as soon as the new stands arrived we gave it a more dignified mounting.

Problem number two was a mysterious rattle inside the Fostex recorder. Removing the hard drive access plate revealed two loose screws that should have been holding the circuit boards down. As it was impossible to replace these without dismantling the machine, we took them out. There were enough screws still in place to keep everything secure.



Removing the hard drive access plate of the Fostex VF16 located the loose screws responsible for a mysterious rattling noise during recording.

The resulting test recordings, made using no EQ or processing, confirmed that we were now getting a very decent basic sound. The acoustic guitar sounded a little more lively with the rug taken up, but as vocals and guitar are often recorded together, we decided the best compromise was to keep the rug down. When vocals and guitar were recorded together, a touch more separation could be obtained by angling the guitar mic downwards and the vocal mic upwards slightly. Although it would have been nice to use figure-of-eight mics to try to get more separation, this luxury was not afforded to us. The only variable pattern mic was the Oktava 012, which has interchangeable heads giving cardioid, hypercardioid and omni responses only. We used it in cardioid mode for this session, aimed at the neck/body junction around eight inches from the guitar (a nylon-

strung c

lassical guitar in this instance). This produced the most acceptable tonal balance, but, as always, I used the headphones while moving the mic to find the optimum position. The vocal/guitar separation was also quite adequate.

When we later tried to record two acoustic guitar players together, we had to move them as far apart as possible, each facing the centre of the room. This reduced crosstalk to an acceptable level, even in this relatively small room. One of the steel-strung guitars was fitted with a type of pickup I'd never seen before and which I was told was originally designed for percussion instruments. The resulting sound wasn't at all bad, so in situations where miking wasn't appropriate, or where there were more instruments than suitable mics, this was plugged directly into one of the line inputs of the Fostex. During our tests, a small amount of reverb was added to the DI'd acoustic guitar so that it better matched the ambience of the miked guitar. Even so, it still sounded better miked, so I suggested DI'ing only when the situation made miking impractical.

One major limitation of the Fostex recorder is that it only has two phantom-powered XLR mic inputs, which meant that only two capacitor mics could be used at any one time, unless the Behringer mixer from the bar was used to provide more mic amps. However, dynamic mics with jack leads could be used in the remaining inputs.



Once an extra mic stand had been ordered, the Oktava 012 was used for recording acoustic guitar parts. Here Murat's collaborator Graeme Bickett has been positioned for recording with rugs being used for rudimentary acoustic treatment.

Recording Hand Drum

Recording a regular drum kit is not practical in Murat's apartment, but most of the time hand percussion (a Turkish darbuka drum) is used. Of the existing mics, I suggested the SM57 might be good for this purpose during a typical live session, though where a spare capacitor mic was available I felt this would give better results because of its superior transient response. For test purposes, we tried the Oktava 219 again, leaving the pop shield in place for convenience. There is generally no need to use a pop shield when recording instruments other than vocals, but leaving it on didn't seem to cause any problems and I wasn't sure about the elastic band situation!

The darbuka hand drum, which usu

ally features a thin aluminium shell, lies somewhere between a bongo and conga in size and can be quite loud. It was quite impossible to record it in the same room as the acoustic guitar without incurring a huge amount of spill so, instead, I decided to try it outside the room in the hallway. During loud passages, the drum caused door panels in the hall to resonate, though very little of this picked up on the mic, which was only around six inches away from the drum. Murat rightly suggested using more rugs to damp these when doing serious recording — recording gear might be scarce in Turkey, but rugs are not a problem! In fact a chair piled with rugs and cushions was placed in the open studio doorway as an acoustic barrier, which, combined



This picture was taken from the main studio room. You can see how Murat has been positioned to play the darbuka hand drum, with a makeshift acoustic barrier to reduce spill to and from the mics in the main room.

Marshall Valvestate combo with a Fender Stratocaster, but it's still far too loud to run in the same room as any acoustic guitars, so we set this up just outside the studio door

or, around the corner and facing away from the studio door. It sounded fine miked with either the SM58s or SM57, but, as we had the Oktava 219 mic already set up in the hallway, I decided to see what kind of sound we could get from using it instead, as lead parts could be added as overdubs when this mic would be available for use.

The mic was set up six inches from the speaker, and initially pointed at the centre of the speaker cone. After making a test recording, we all felt that the result was rather bright, so I suggested moving the mic to one side of the speaker to warm up the tone, and also adding a little more low end on the amp EQ. Graeme said he'd been told that moving the mic to one side created a brighter sound, but we soon confirmed that the opposite was true (it's nice to know the same laws of physics apply in Turkey!), and with the mic aimed at the edge of the speaker cone we got very close to the way the amp sounded in the room. The main difference between using the Oktava and one of the dynamic mics was that

with the few extra feet of distance, gave us far better separation than we'd expected. When soloing the acoustic guitar track, the drum was now barely audible. The guitarist was playing in the usual position with his back to the window, facing the open doorway, so there was good line of sight for communication between the two players.

Bass & Electric Guitar

Murat has a Carlsbro Electroacoustic amp in the studio room, so I asked him if we could try this for recording bass. As it turned out, it produced a really nice bass sound, but attempts to DI it from its slave output resulted in a lot of background noise, so we finally decided on miking it around six inches from the speaker. Miking produced a really warm, full bass sound with bags of definition and far less noise, but of course the bass spilled onto everything, so the amp had to be relocated to the adjacent lounge, where the amp was aimed at a soft sofa to try to minimise reflections. Because of the lack of headphone monitoring, the bass player had to stand between the amp and the studio door to hear both his own playing and that of the other musicians, but the solution was workable and provided excellent separation.

Electric guitar, and occasional bass, is usually furnished by Graeme Bickett, a Scottish friend of Murat's who plays guitar in the bar band during the summer season. Graeme uses a small



The bass cab was positioned in the lounge, away from the main studio, and pointed towards the sofa to reduce reflections.

the higher harmonics sounded more lively, helping to play down the usual close-miked sound.

All guitar players have to play at a certain volume to get a good tone, even with a master volume amp of this kind, so I raided one of the sofas for cushions to improvise an isolation box around the amp and mic. Rugs were then draped over the top for further isolation, but we left a small gap facing Graeme, so that he could hear what he was playing. Spill from either the bass or electric guitar to the miked acoustic guitar was minimal using this arrangement, and close to what I've heard in some professional recordings from big studios. Where the electric guitar was being recorded at the same time as the drum, the close miking of the drum avoided excessive spill from the guitar amp, provided that a minimum distance of four to five feet was maintained between the drum mic and the guitar amp, and that the drum mic wasn't directly in front of the guitar amp.

These measures dealt with the spill issues very effectively and, as long as the bass and guitar players chose positions where they could hear their amps and the other instruments, the solution was perfectly viable. However, I suggested that extra headphones and

a headphone distribution amplifier should be placed near the top of the 'what to buy next' list, as this would enable all the players to hear a reasonable balance regardless of where their amplifiers were. This could be fed from the headphone out of the Fostex or from a pre-fade send.

Working With The VF16's Processing

Murat told me he'd had bad experiences with sound engineers applying excessive compression to his voice during live performance, so he was understandably suspicious of using it. He prefers to use mic technique to control his vocal levels, so I came up with the compromise of trying a combination of mic technique and mild post-recording compression, courtesy of the Fostex effects processing. I also felt that, while acoustic guitar can sound very good recorded 'as is' in a live environment, the vocals would work better with a suitably tweaked reverb setting, again using the internal effects of the Fostex.

We ended up using a compression ratio of between 4:1 and 6:1 for vocals, and adjusted the compressor threshold to produce sufficient gain levelling, but without causing audible side effects. The default attack and release settings were left as they were. Adjusting the amount of compression was quite difficult, as the VF16's compressors have no gain reduction read-out, so everything had to be done by ear. Also compression is only available on four of the VF16's channels, and can't be used at the same time as EQ on those channels.

For reverb, Murat wanted a fairly natural, warm sound so we ended up using the Old Plate preset and then adjusted the decay time to 1.8s and dialled in 57ms of pre-delay. Reverb was added mainly to the vocals, but was also used with the DI'd acoustic guitar to compensate for the lack of an acoustic environment. Enough



For recording electric guitar parts, the Oktava 219 was positioned a little to one side of the speaker cone, with a makeshift booth around it, constructed from cushions, to reduce spill. The positioning of the cat was, of course, critical to the final sound...

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Published in SOS November 2002

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Studio SOS

Noor Ali

Published in SOS December 2002

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Studio SOS

Hums, buzzes and noise were stopping Noor Ali recording his guitar parts, so SOS headed over to Worcestershire to set things straight.

Paul White

Noor Ali is a guitar player with some history of recording in commercial studios, but he's since decided to buy himself a little multitrack setup and record his songs at home. He's a regular SOS reader and says it's helped him a lot with understanding recording techniques, but he called us because he was having problems recording his electric guitar. He felt that his system was producing too much hum, and, as so often turns out to be the case, the reality was that a number of problems were acting together.



Untangling A Complicated Hum Problem

Noor's studio is set up in one corner of a downstairs room in his Worcestershire home, and is based around a Zoom MRS1044 10-track recorder and a Joemeek VC3Q voice channel. Noor has a Joemeek JM27 capacitor mic and currently monitors through headphones, only checking the result on his hi-fi system after mixing to Minidisc

. The studio is home to a number of guitars, but most of our tests were made with a Gibson SG fitted with humbucking pickups. There are two guitar amplifiers, a 2 x12 Sound City combo and a smaller Laney 15W combo, both all-tube designs. Trying the Sound City first confirmed that it had a bad hum problem, even with no guitar connected, so I suggested that he get it serviced. There are numerous factors that can cause hum in valve amplifiers, including worn valves, incorrect output valve biasing and an imbalance in the heater voltage (most have a centre tap to ground). Furthermore, on an amplifier the age of this Sound City, any number of resistors could have drifted out of spec and time doesn't treat capacitors at all well either.



Noor Ali's studio is based around a Zoom MRS1044 multitracker and Joemeek VC3Q. These units both use mains adaptors, and Noor was monitoring via headphones, which meant that there was no ground connection at all in the system — one reason for the hums and buzzes he was getting while recording guitar.

That meant we had a choice of recording via miking the Laney or DI'ing using the Zoom's built-in effects. We decided to check out the miking method first, and I was interested to see what miking arrangement he normally used. It turns out he'd been angling the mic downwards quite steeply and placing it close to the centre of the speaker, which had the effect of softening the tone slightly because the mic was being used off axis. I felt we might get more repeatable results if we went back to the more usual setup where the mic points directly at the speaker but is then moved outwards towards the edge of the cone if a warmer sound is needed.

After setting up the mic stand, we powered up the amplifier and plugged in the guitar. Sitting close to the amplifier gave the predictable hum problem, as the guitar pickups and wiring coupled into the hum field generated by the power transformer in the amplifier — even good humbucking pickups can only do so much — but by increasing the guitar/amp separation to around five or six feet, the situation was made better. The level of hum also depends on the angle of the guitar relative to the angle of the amplifier, so I rotated the amplifier to find a null and ended up with it being almost side-on to the guitar.

Once a good position had been arrived at, the level of hum and buzz was adequately low when the guitar strings were being touched, but became quite loud when the strings were released. This is often indicative of less than optimum screening within the guitar (around the controls and switches), but a practical workaround was to use a conductive wrist strap connected to the bridge of the guitar using a thin flexible wire and crocodile clip. Commercial variants of this idea can be purchased from UK electronics suppliers such as Maplin Electronics, where they are used b

y service engineers to prevent electrostatic build-up when working with electronic equipment that is sensitive to static electricity. Having proven the principle, we went on to do the rest of the recording with Noor making sure he kept one hand in contact with the strings or bridge at all times.

Aside from a little amplifier hiss, which wasn't loud enough to be a problem, we now had a working amp and guitar set up, though I wasn't at all happy with the overdrive sound the amp was giving — it was hard, unyielding and had a rather nasty barking quality that isn't normally a characteristic of Laney amplifiers. On discussing this, it turned out that Noor had changed all the valves in the amplifier (originally Russian valves that were easily driven into distortion) for some highly specified replacements, but I suspect that the biasing is less than optimum for these new valves, so the amplifier really needs to go in for a service. Noor had also upgraded to a different speaker, but that didn't seem to be giving any problems. The amplifier sounded quite well behaved when the gain was backed off to clean up the sound, so I suggested we use his Boss OS2 overdrive pedal in conjunction with the amplifier running fairly clean, and straightaway this produced a more comfortable 'British blues' sound.



Recording Tests

At this stage, I asked Noor to make a test recording using his normal working methods, and on playback some unpleasant clipping distortion was immediately evident, even though he'd set a sensible recording level on the Zoom MRS1044. On checking the signal chain, it transpired that the clipping was taking place in the VC3Q preamp — the VC3Q has more headroom than many designs, but in this instance it was all being used up! The Joemeek mic is a sensitive small-diaphragm capacitor model, but neither it nor the VC3Q are fitted with pad switches, so the signal level entering the preamp was very high. I had to turn the input gain to its absolute minimum to get the level down to manageable proportions, and even then it was clipping the output stage on peaks because Noor had added a little EQ boost, and that was enough to push the signal past the limit.

One solution would have been to use EQ cut rather than boost to achieve the same tonality, but as I was starting from scratch I felt it would be better to zero the EQ settings, then use the mic position and amp controls to adjust the tone. Backing the mic off to around four inches from the speaker and moving it toward the edge of the speaker left us with a good sound and just enough headroom in the VC3Q to avoid clipping, even in situations where a little compression would be added. Noor had been using compression routinely, so I explained that, while compression can help even out the sound, every 1dB of gain reduction translates to 1dB increase in background noise when you're not playing. This exaggerates any hiss or hum present in the miked sound, so setting as little compression as you can get away with is safest.

On repeating the recording test, we were pleased to discover that what played back from the Zoom was very similar to what we were hearing from the amplifier. All the recording needed was a little of the Zoom's internal reverb while mixing, as the Laney amplifier didn't have a spring reverb. At this

stage we felt we had established a valid method

ethod for recording the guitar that sounded right and didn't include an unacceptable amount of hum, buzz or hiss, though Noor demonstrated that if he touched any metal plugs connected to the Zoom recorder, or if he touched the metal base of the unit, there was a nasty buzz in the headphones. He explained that this caused a serious problem if he tried to DI the guitar via the Zoom's internal effects section rather than miking an amplifier, though he felt that even if the buzz were cured, he'd still record using the amplifier, as it gave a more authentic blues/rock sound.



Noor's Laney amp couldn't provide the 'British blues' sound he was after on its own, so the gain was backed off on the amp and overdrive was provided using a Boss OS2 pedal instead, giving a much more suitable tone.

A quick look over the system revealed the source of the problem. The only two mains-powered items were the Joemeek VC3Q and the Zoom MRS1044, both of which run from mains adaptors that don't carry the mains earth through to the units themselves. This lack of a hard earth can help avoid ground loops, because, in a typical setup, the monitor output of the Zoom would be connected to an earthed monitor power amplifier, which would provide a central ground connection for the whole system. However, as Noor was using only headphones, there was no earth connection anywhere in the system, which meant the metalwork of the VC3Q and Zoom MRS1044 were effectively 'floating'. This was confirmed by using a piece of wire to make a temporary connection between the metalwork of a jack connected to the Zoom MRS1044 and the mains earth — the buzz problem disappeared instantly. For a more permanent solution, I suggested the option of fixing a tag washer to one of the screws in the base of the MRS1044 and earthing that, but pointed out this wouldn't be necessary if a monitor amp and speakers were added. Noor had already realised that the headphones didn't provide an accurate impression of bass, so was considering buying a monitor amplifier and a small pair of reference speakers.

A Good Clean Job

As is so often the case, what started out as an apparent single problem turned out to be a combination of factors, but, once these were addressed in a logical order, it didn't take long to identify them and come up with solutions. As with any recording, a good result starts out with a good sound at source, and around half our effort was focused on achieving that. Had the amplifier been working properly, that particular job would have been a little easier, but amp position, mic position and quality of guitar screening all played their part. On the recording side, it was only when we'd sorted out the system grounding and unusual gain structure that we started getting really good results. However, the whole process took less than an hour and made a world of difference to the recorded result. **SOS**

Glossary

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Published in SOS December 2002

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