

Get organised



Setting up your computer might not seem like the most exciting mixing project you'll undertake, but do it properly from the start and you'll be free to focus on the important thing: your sound

You already know that your computer is the most powerful musical tool available to you in your studio. But with that power comes more complexity than you'll ever need, and – dare we say it – some responsibility, too...

Mixing can be simple or it can be complex, but either way, it can also be quite confusing. And that's the last thing you want when you need to be focusing on your sound. Avoid getting into a pickle and give yourself a head start by taking

some time to arrange things in a clear manner. Then, if you do need to make a U-turn, you should be in much better shape.

So what's the best way to set up your computer for mixing? Taking a quick look at an analogue mixing console is a good starting point. A well set up mixing desk will be configured differently for recording and mixing, with good reason. At the mixing stage, tracks and inputs are routed via the line inputs (not the monitor inputs), allowing you to make full use of the

channel processing. All signals, including any auxiliary effects returns, are then bussed via a master fader into a stereo output. But on many desks you'll also find that the group busses you used to send signals to your recorder during the recording stage can now be re-patched, allowing you to create stereo sub-mixes as well.

So how does that translate to mixing in your computer? Well, as long as your software has good bussing options (the full versions of Cubase, Live, Logic, Sonar and Pro

Tools all do), setting up your computer in this way can be a real bonus. Not only does it keep things tidy and easy to understand, it should also make better use of your CPU. Fundamentally though, both of these aspects will help you achieve better mixes – and that's the important bit.

Gain structure primer

Although big mixing desks may look incredibly complex, in reality they're actually quite

STEP BY STEP Limiting



1 It's extremely easy to get into the habit of overloading channels, and often the clipped sound can act just like limiting the signal. The problem is, it messes up your gain structure. Here, simply overloading one channel has caused the master output to clip as well. >>



2 It may seem obvious, but if you do want to clip or limit signals, there are often better ways to do it. Here we've selected a simple limiter plug-in, slammed the signal up to zero to achieve the desired clipped sound, then turned the whole thing down so that our output doesn't overload at all. >>



3 You may also find that certain plug-ins include clip protection. In Logic's standard compressor, for example, hidden away in the extra parameters are some overload options (**Soft**, **Hard** and **Clip**). As it happens, none of these will actually let you overload the channel, which is rather handy.

simple. Take a look at the bottom right of this page, where you'll see our block diagram of the signal flow for one channel of a mixing desk as it operates at the mixing stage of a project. As you'll see, it's all pretty straightforward. What's interesting, though, are the various points of gain control. These may be amplifiers (plus and minus) or simply trims (just minus), but the point is, every channel has them and they're very handy indeed.

Now, thinking back to how most software is set up, it's likely that most tracks will simply be routed straight to the main output via a level fader. What's more, that fader will most likely be accompanied by a peak meter (we'll delve more into metering further on).

It's all good stuff, but imagine you're blending 30 or more tracks to that single stereo output. The chances are, before you know it, it'll be flatlining good and proper, and not only will you have run out of bits, you'll also be distorting your signal. As it happens, most software will overload to a degree without sounding too bad (and will sometimes indicate the overload amount). The problem is, this can lull you into a false sense of security.

So what's the answer? Well, anyone used to mixing on analogue boards will be fully au fait with trimming back the faders to stop the output buss overloading. But if you really want to stay in control, one solution is to build various additional gain layers into your mixing setup. These could be simply a couple of sub-groups (such as backing track

“IT'S YOUR OUTPUT METER THAT YOU WANT TO BE TICKLING THE ZERO POINT, AND THIS MEANS MOST OF YOUR INDIVIDUAL TRACKS WON'T BE METERING ANYWHERE NEAR ZERO”

and vocals) or a much more complex setup with extra gain trims all over the place. Whichever route you take, you'll then have a safety net should things get out of control, and it means you won't be mixing with all the faders pulled down to where the scaling gets less accurate.

There will be some people who say they like to clip the levels, as this acts as a form of hard limiting, and that increases the energy of sounds. This is fair enough, but if that's your bag, there are other ways you can achieve it – using a genuine limiter plug-in, for example – which won't result in the levels overloading further down the signal path. Plug-ins such as Steinberg's Loudness Maximizer or Logic's Adaptive Limiter allow a choice of ways to squash the signal up to zero, and you can then turn the overall level down without losing the effect you want.



STEP BY STEP Metering



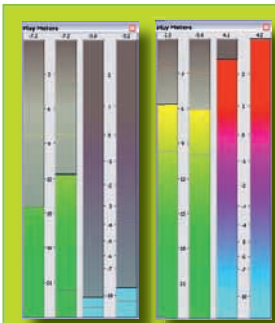
1 In this simplified mixing setup, there are three audio tracks, one software instrument and one reverb routed directly to the output buss. Looks pretty straightforward, and all signals are metering within their margins. And yet, the output buss is already peaking in the red. We've used Logic Pro here, as it gives us some feedback about just how far over the top we are. >>



2 For the purposes of demonstration, we've turned each signal level up by a couple of dB, and immediately some of the channel faders are going into the red. The master output is now seriously in the red, and metering 6dB over. This is the maximum overload indicated in Logic – any further level won't actually show on the meters. >>



3 Turning our faders up another couple of dB pushes each channel further into the red, but our master fader is still only showing 6dB over, so now we're really in the dark. Moving the master fader down 6dB emphasises the point, as the buss is still in the red. All this goes to show that when you get into overloading channels and outputs, the metering can become quite misleading.

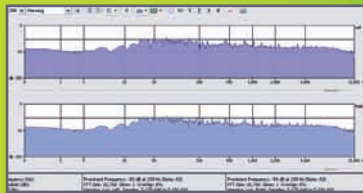


PPM vs VU

Rather than rely solely on peak meters, it can be good to see things on VU meters too, as they emphasise different things. Here we've got a meter that shows peak levels on the left and VU on the right. The first one is from a passage without too much low frequency content. The second is at the peak of a crescendo. Although the peak meter differs between the two, the difference for the VU is more marked, indicating a big difference in the loudness of the track.

Spectrum analyser

Another useful tool is the spectrum analyser. These can differ considerably in style, from real-time graphs to full track sonograms. A basic spectrum analyser can be very informative about the overall frequency content of a track, and can also be useful for comparing your own work to commercially available music you like, helping you to develop your own sound.



Metering

You'll encounter two main types of audio level meters: PPM (peak program meter) and VU (volume unit). Typically, the needle meters you see on a desk are VU meters, while the metering within your software will tend to be of the peak variety. This is for good reason, as in digital audio your main concern is making sure you don't go over zero, while still trying to use as much of the dynamic range as possible. However, when it comes to the mixing stage, if you're going to successfully buss those signals

together without overloading the outputs, you'll need some headroom. That means turning some things down.

In practice, there are a few things to bear in mind. Firstly, it's your output meter that you want to be tickling the zero point, and this means most of your individual tracks won't be metering anywhere near zero.

Secondly, peak meters are best at indicating transients, but with sudden sounds, even the fastest computer can sometimes struggle to keep up. Ultimately, the meters are there to help, but don't rely on them solely.

Thirdly, track down a VU-style meter and use it on your mix output. It'll give you a much better indicator of the perceived loudness contained in your mix.

Finally, many software packages include spectrum analysis plug-ins. Not only do these look great, if your monitoring is limited they could also become your salvation. Take the time to run some commercial mixes you like the sound of through one of these so you can see what's going on.

Layout

Unless you're lucky enough to own an enormous hardware controller to drive your software from, chances are all the visual data – from faders to waveforms – is going to be crammed onto the same screen. Thankfully, each software package has its own way

of dealing with this, although some are more flexible than others. However, whether you're using Logic Pro's complex screensets, the simpler two-screen setup of Ableton Live, or something in between, your mixing experience will improve considerably if you arrange the tracks in logical fashion.

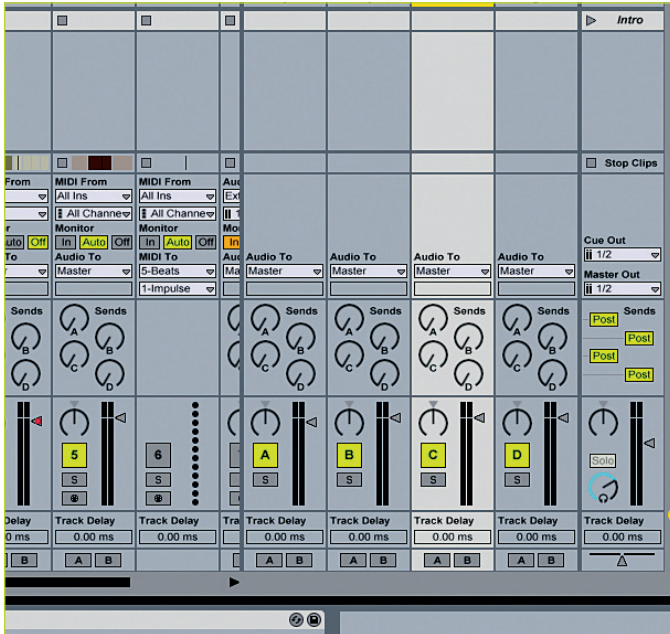
Putting down tracks quickly during the production process can leave you with an arrangement that's less than clear. At the simplest level, you should at least shift the tracks around so that similar elements are close together. If your song is very complex, take the time to sort things properly. If you have the option, it's worth colour coding individual tracks, or elements within them.

Equally, keep your auxiliary returns together and select your auxiliaries so that they follow an obvious pattern – with, for example, delays on the first four auxiliaries in order of length, followed by reverbs using the same order. It may look like a number-crunching approach, but if you always follow the same rules it'll save you a lot of time, and if you're mixing with other people in the room, it'll make you look like a pro.

The other thing to always keep a look-out for is opportunities to share plug-ins. Using auxiliaries for delays and reverbs is the traditional example, and this is far more CPU-efficient than using these sorts of effects as inserts. But you can apply the same theory to sub-mixed elements, which can then share compressors, EQs and the more CPU-hungry limiters.

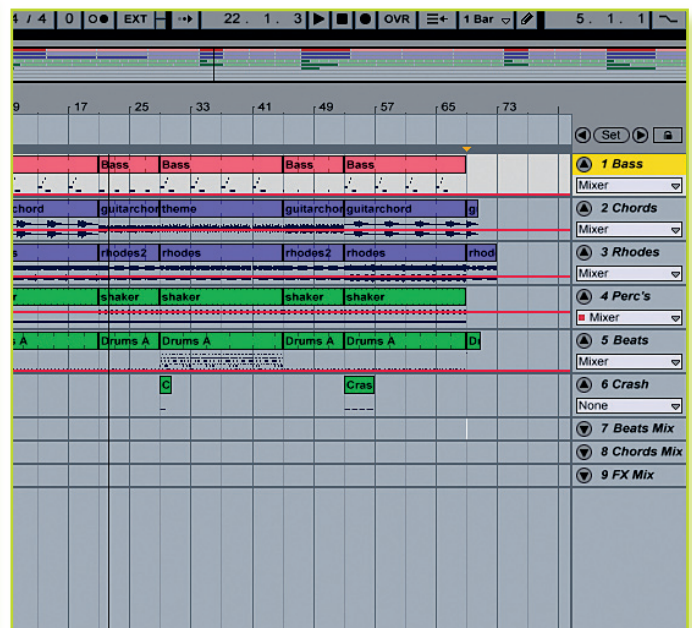
Setting up

FX allocation



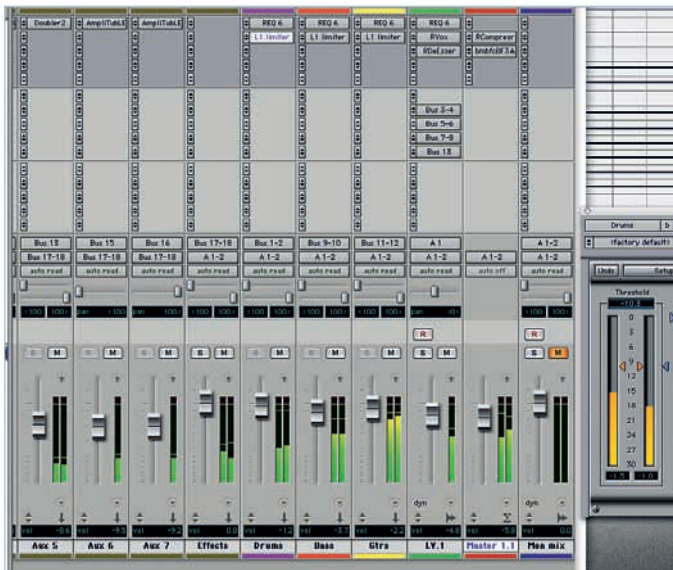
If you want to keep things simple, start by laying things out in a logical fashion. In this basic Live session, we've set up four auxiliaries in order – eighth delay, quarter delay, short reverb, long reverb – and labelled them accordingly. Now applying the auxiliary effect we want shouldn't be confusing. >>

Colour code



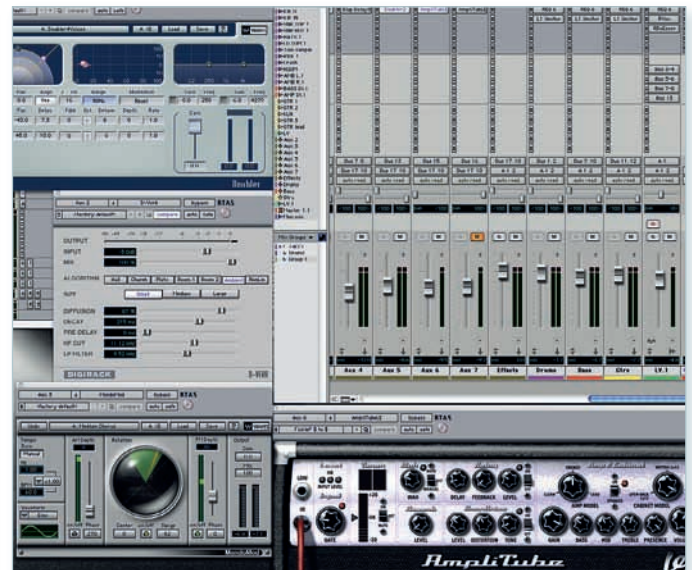
When programming or recording your track, it's often useful to colour code sections according to their content (verse, chorus, drop section and so on). At the mixing stage when the track structure is established, this is likely to be less useful, so switch the colour coding to emphasise broader elements in the track. Here we've used colours to divide our track into three simple categories – beats, bass and chords – and visually, things are much clearer. >>

Multiple sub-mixes

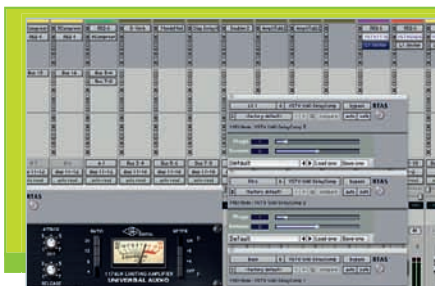


Once you get into the concept, it's quite simple to set up multiple sub-mixes. Here we've stuck to a simple bass, drums and guitars split, with the auxiliary returns also grouped together. Each sub-mix is being processed, and all have maximising limiters across them, although the attenuation is subtle. With the limiters keeping the sub-mixes in check, the master output can be treated to subtle compression. >>

Auxiliary effects



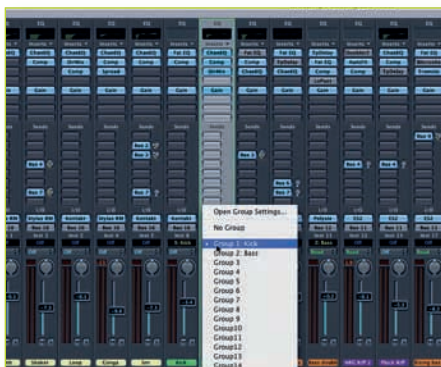
Some plug-ins will gobble up all of your CPU, so if you think you're going to be using them on a number of tracks, put them on auxiliaries. Here we've got a bunch of powerful plug-ins on auxiliary sends, which save us enough CPU power to stick those equally powerful maximising limiters over our sub-mixes.



Delay compensation

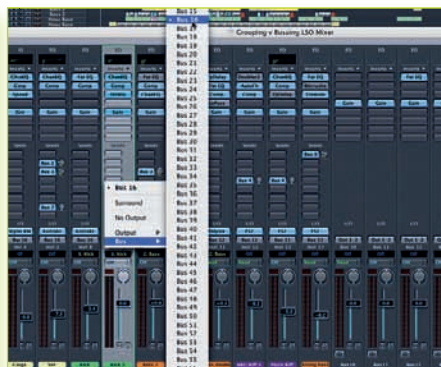
Some software still requires you to make manual adjustments for delay compensation (see *Pitfalls*, over the page). If you're using sub-mixes, you may want to keep a close eye on this. In Pro Tools LE, using powered plug-ins, we've bussed all elements of our mix to sub-mixes. This leaves us free to use a combination of those powered plug-ins and delay compensators to keep everything in line. >>

STEP BY STEP Bussing and grouping



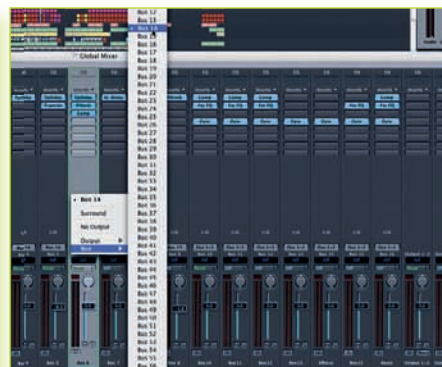
Simple groups

The terminology tends to vary between software packages, but on the whole, grouping refers to the linking of different tracks. Here we've taken two kick drums, set their levels, then linked them. We've then done the same with the bass parts. »



Simple buss

We've set up a number of extra busses so that we can sub-mix certain elements of the track. Our first sub-mix will handle beats. You'll also notice that we've taken the precaution of trimming the level down on the buss, using a gain plug-in. »



Effects buss

To keep things tidy, we've also decided to group our effects returns to one buss. This can change a number of uses during the mixing process, including general increasing, decreasing and muting of overall effects levels.

“MANY SOFTWARE PACKAGES INCLUDE SPECTRUM ANALYSIS PLUG-INS. IF YOUR MONITORING IS LIMITED, THEY COULD BECOME YOUR SALVATION”

Buss and group

We've already mentioned that bussing signals into sub-mixes will help both the organisation and sound of your mix. But how does that differ from grouping? Typically, grouping involves linking together a bunch of existing tracks so that they act as one unit. In Logic and Pro Tools, you can simply create specific groups that link tracks for editing, automation or both. This way you could, for example, set up a group for all drum kit or 'beat' elements. Then whenever you change the level of one element, all the others will follow.

Bussing, on the other hand, refers to the concept of blending certain elements to form a sub-mix. The difference is that the sub-mixed signal can then be processed as one, whereas grouping merely allows you to control a bunch of tracks. Of course, there's nothing to stop you doing both.

So why is this significant? Well, one of the main considerations is your auxiliaries. If each of your grouped tracks is sending a level to an auxiliary (and typically the level

is affected by the fader level), then changing all of your fader levels together (using a group) will change the send levels, but in a consistent manner (for all your grouped tracks). However, with a bunch of tracks that are simply bussed into a sub-mix, changing the overall level of the sub-mix won't affect the send levels of the individual tracks routed to it, so the send levels remain constant.

It may sound like a pedantic point, but in practice it's a significant one, especially if you need to modify your group levels and maintain the relative level of the auxiliary returns. One alternative is to buss your effects returns to their own buss too.

Pitfalls

OK, we've already touched on one of the pitfalls of setting up complex sub-groups, and that is, you need to make sure you can confidently modify your groups or sub-mixes without affecting your auxiliary effects balance. There are a few other things to consider as well...

Firstly, delay compensation. If you sub-mix tracks and use plug-ins across them, you will inevitably introduce a degree of processing delay. If your software has full automatic delay compensation, then this shouldn't be a problem, as everything will be delayed to match the slowest plug-in. However, not all software uses this, so you may have to think about using manual delay adjusters. This will be particularly apparent if you use powered plug-in systems such as the UAD-1.

The second trap is that you can

end up with so many sub-mixes and groups that it defeats the point of setting them up. Your best bet is to go back to basics: sub-mix your backing track, vocals and effects.

One last thing...

And finally, before you think that it could all be a bit of pain, we have one word for you: automation. If you like to get busy with automation, you'll be only too aware that once you've done lots of level adjustment, rebalancing elements while retaining your automation can be a fiddly process. With a few sub-mixes, it can be much simpler. **cm**

