Ardour: Equalise

PART 3: Our audio guru **Daniel James** covers mixing your track, mixer automation, and effects processing with LADSPA plugins.

Our expert

Daniel James is a co-founder of the 64 Studio distro, and a long-time user of *Ardour* for recording projects. He can keep up most of the time, if you tell him what the chords are first.

I n the first part of this tutorial in **LXF101**, we looked at installing *Ardour*, setting up Jack, and recording our first track. Last issue, we recorded more tracks for bass, synth and guitar. We also learnt to use the click track, had an initial try at editing, and found out how to export a session. If you've burned your track to CD and tried it on your stereo, unless you are a naturally talented sound engineer, you will probably have found that it sounded quiet, and in subjective terms, lacked clarity and punch. This is most likely because you are comparing it with the commercial releases that you are used to hearing on the radio or online, which have gone through a lot of sonic mangling to make them sound that way. Since the 1950s, popular music producers have used ever-greater amounts of analogue and digital processing to make their recordings stand out from the rest.

The maximum possible level of digital audio equipment is referred to as 'zero decibels from full scale', or 0dBFS. Levels below this maximum level are therefore expressed as negative numbers, which you can see displayed on the *Ardour* meter strips in the mixer.

❯ *Ardour*'s mixer scale is calibrated from +4dBFS all the way down to -50dBFS.

In theory, any signal that exceeds 0dBFS ('going into the red' on the *Ardour* meter) should introduce clipping – a nasty-sounding digital distortion wherein the top of the waveform is clipped off, turning a sine wave into a kind of square wave. This isn't the same as the analogue distortion that you get from a guitar amp (think Stevie Ray Vaughan and his overdriven Fender amps) or tape machine being overdriven; digital distortion is nasty, and tiring on the ear. So we want to avoid it if at all possible.

In fact, *Ardour* and Jack can forgive 'overs' as long as they aren't too extreme, but you have to take care of these before you finalise a CD, or produce an Ogg Vorbis file for your website. This is because the export process, and the playback equipment, isn't at all accommodating when it comes to wave clipping. The meter on

❯ Note that although the meter reads around -20dBFS, the number at the top of the meter indicates there has previously been a peak of -4.5dBFS. The small orange bar at -10dBFS tells us the most recent peak level.

> ## "Digital distortion is nasty, so we want to avoid it if at all possible."

the *Ardour* mixer's master bus strip is the one to watch for overall output level, and like the the other meters, it will show the highest peak level reached during the session as a numerical value. This is useful just in case you miss a transient over, too short in duration to hear at first. These transient overs are often caused by drum tracks, and percussive sounds generally.

Ye olde recordinge methods

Back in the days of studio tape machines, it was the fashion to run the recording meters 'hot', which means to go as close to the maximum level as possible for each track. This was due to the inherent background noise of analogue tape formats, which meant that quiet recordings had more hiss, once they were boosted up to maximum level. You may recall similar advice if you were in the habit of making cassette compilations back in the 1980s – to keep the meters in the red. With modern digital recordings, there is usually no need to do this. The background noise level – sometimes referred to as the noise floor – is very low in any half-decent audio interface designed for home studio use. This means you can record and mix each track without the meters glowing red

❯❯ **Last month** Recording guitar, bass and synth… Now go form a band!

sound levels

all of the time, and leave raising the average level to the mastering stage, which we'll be looking at in more depth next issue.

The best way to record and mix is to leave yourself some headroom between your peak signal and the 0dBFS point on the meter, because once you've manipulated the track with fades and effects, it may well be coming back louder. Remember though that you can't improve signal to noise ratio by boosting level in the mix, because you boost the noise on the track at the same time. We're aiming for a track that's recorded loud enough to mask background and equipment noise, but with at least a few decibels of headroom at the top of the meter during the mixing stage.

Finding the balance

Start Jack and open the *Ardour* session that you created previously. Hit the transport rewind and play buttons, then switch to the mixer view using Windows > Show Mixer, or the ALT+M keyboard shortcut. You should see the meters flickering, with two green/yellow-coloured vertical bars, one each for left and right channel output. This is because the meter is in Post mode by default, which shows the Post-fader playback level, after any adjustment made in the mixer strip. You can switch modes between Pre-fader, Post-fader and Input monitor level using the small grey button below the meter. If you click on the Record button on any particular track, the meter will switch into Input mode automatically.

As the level increases towards 0dBFS, the meter bar colour changes to orange, and then to red if there is an over. If this happens, the peak level box above the meter strip changes to a white number on a red background, to alert us to the fact that we have a potential clipping problem. A mouse click on this peak level box will reset it to the current peak value.

Ardour's mixer can be used with a digital mixing desk or a control surface, but if you're just starting out, you can adjust levels with your mouse. If you click on the light grey bar to the left of

> **We're running too hot here – a session mixed like this is going to sound pretty rough.**

each pair of meter bars, and hold the button down, you can move the fader up and down with mouse movements. By default, each fader begins at zero decibels, 0dB, which means that no boost or cut is being applied to the signal. You can see the number in the box at the top of the fader change, as you move your mouse up and down. A single click on the fader will bring it up or down in small easy-to-control increments, depending on whether you click on a point above or below the thin white 0dB line.

> **A stereo track is shown on the left, featuring hard panning. On the right is a mono track, panned to the centre.**

Balancing levels in the mix isn't just about making sure there are no overs. There are subjective and aesthetic decisions to make too. For instance, in some styles of music, the drums are meant to sound like they're off in the distance, which corresponds to a lower level in the mix. In other genres, they're right up front. The only way you can get this right is by practice, trying out your mixes on different playback systems until they sound good to you. Friends and family will also volunteer their advice, which will probably be negative at first: "The guitar is too loud", "I can't hear the singing", and so on.

Most live instruments and vocals are recorded with mono microphones, yet the output format is usually stereo. While it's possible to produce a retro all-mono record, like Seasick Steve's 2006 album *Dog House Music*, most recordings since the mid 1960s have featured a 'fake stereo' sound field, created by panning individual tracks. *Ardour*'s panning control is towards the bottom of the mixer strip, and features a thin green line that you can click and drag to either side of the stereo field. You can also click on the small triangles at the top of each panning control to select hard-left, centre or hard-right panning automatically. For stereo tracks, there are two controls, with channel 1 (left channel, by default) at the top. You can drag the left channel to the hard-right pan, and vice versa, if that takes your fancy.

There are some conventions for panning in popular recording, which approximate to vocals in the centre, guitars and bass panned to left and right. Live drum recordings are usually panned »

» **If you missed last issue** Call 0870 837 4773 or +44 1858 438795.

» to the rough positions of the microphones on the kit. It's not like you will be taken away by the mixing police for breaking these rules, so experiment if you want to. One thing to watch out for is using too much panning. This can leave a 'hole' in the middle of the stereo image, if you're not careful.

Mixer automation

Balancing the tracks in a mix works best when it's done dynamically – making the lead guitar louder during a solo would be a classic example. So far, we've been controlling faders and pans manually. In the analogue days, this was the only way to mix, and sound engineers would have to adjust controls in real time, while the mix was being dubbed to a second tape machine. Hand-written notes were kept for the benefit of engineers who might have to remix the material later, but these were often lost. Since we're using a computer, we can have the software record our mix decisions, and play them back later; this is known as mixer automation. *Ardour* saves mixer automation data in the session directory, so in theory we should be able to recall exact mix parameters many years later, even when the original mix technique has long been forgotten.

Ardour's fader automation is controlled using the small grey button directly beneath the fader. By default, it has the letter M for Manual as a label. Click on this button, and you'll see the other options are Play, Write and Touch. Write and Play modes are pretty obvious – you set Write mode when you want to record moves on an individual fader, and set Play mode to recall them. Try it now, and you'll see that as the red playback cursor in the *Ardour* editor moves to the point on the timeline where you wrote a fade, in Play mode the fader will move in a ghostly fashion. You can't make manual adjustments to a fader when it's in Play mode.

Touch mode lies somewhere between Write and Play modes, in that existing automated fades are played back, but if you touch the fader, new automation data will be recorded. The touch mode is handy if you're happy with most of the fades you've already written, but you need to make some adjustments here and there.

Pan automation works in exactly the same way as fader automation, and is controlled independently by its own Play/

> "Too much panning can leave a hole in the middle of the stereo image."
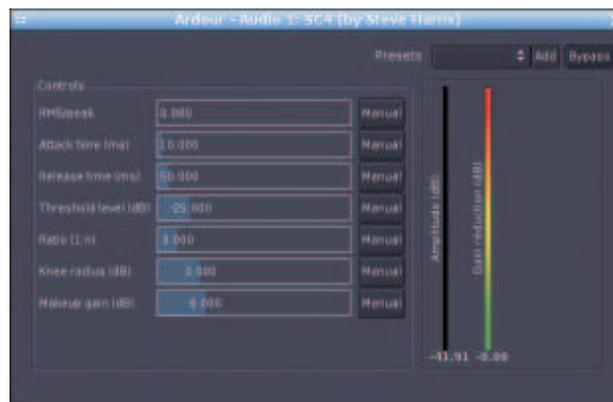
❯ **Some typical settings for the SC4 compressor. The Gain Reduction meter is showing zero, as the plugin is waiting for the next loud segment of the music we're editing.**

Write/Touch button. This small button is on the right side of the mixer strip, just below the pan controls.

Compression, limiting and equalisation

Now we've got the tracks in balance, with fades in all the right places, it's time to address three common mix problems:

■ Some of the tracks are at an inconsistent level, and it would take forever to sort out with fader automation. (This is often the case with vocal tracks).

■ Some transient bursts of sound are pushing the meters into the red, even though the rest of the track is quiet.

■ Many of the instruments have a muddy, boomy sound, so it's hard to hear them individually.

Sound engineers like to attack these problems using the triple whammy of compression, limiting and equalisation. These are traditional techniques used since the early days of rock 'n' roll, when the aim was to make recordings sound more exciting when played on a jukebox. Fortunately for us, these methods translate well into the digital domain, and there are Free Software plugins
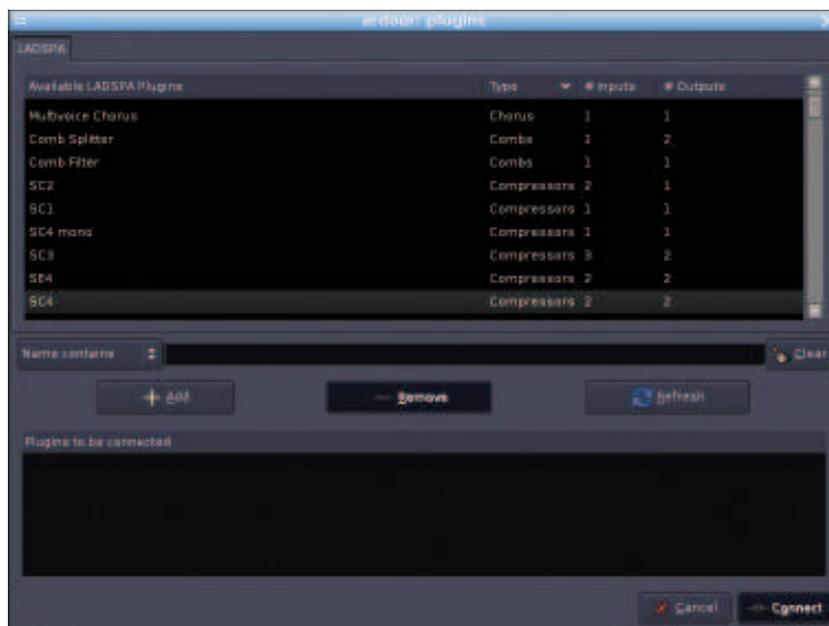
❯ **Ardour's plugin selection dialog, featuring a broad range of LADSPA effects.**

> **Two settings for the Hard Limiter plugin. Increasing the Limiter and Residue Level controls softens the effect.**
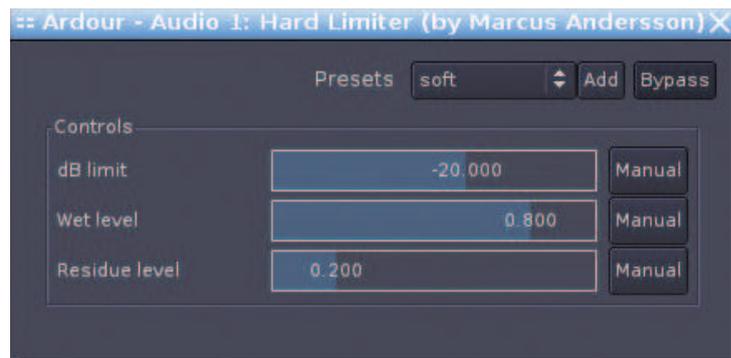
for *Ardour* that cover all these effects and many more. LADSPA has been the audio plugin standard on Linux for several years, and the API of its successor, LV2 (LADSPA version two), was only finalised in January this year. For the time being, you're likely to find that it's the original LADSPA versions of the plugins that have been packaged for your distro of choice, and we'll be using the *swh-plugins* package (available from **www.plugin.org.uk**) in the following examples.

Compression is the tool used to solve the problem of inconsistent sound level. The way it works is that every time the signal passes a pre-determined threshold, the compressor plugin applies a specified gain reduction on that part of the audio. The effect is to make the loud parts a bit quieter, while the quiet parts are left alone. This creates more headroom, which is why a compressor often includes a boost control – raising level after the compression stage, for gain with reduced risk of clipping, but an increased noise floor.

In *Ardour*'s mixer, right-click in the black box above the Mute and Solo buttons for the track you wish to compress. This is the pre-fader plugins box, and there's a corresponding Post-fader Plugins box further down. Select 'New Plugin' from the top of the pop-up menu, and a LADSPA plugin selection dialog should open if your plugins are correctly installed. (The same right-click menu can be used to delete a plugin once you no longer need it). Plugins are listed alphabetically by default – you may find it helpful to click on the Type button, which arranges the plugins by their function. Under the type Compressors you should find SC4, which is a stereo in, stereo out plugin. There is an alternative SC4 Mono compressor available for tracks with a single channel. Select SC4, click on the Add button, and then the Connect button to close the dialog box.

The gain game

The name (SC4) should now be visible in the Pre-fader Plugins box of the mixer strip. The brackets mean that the plugin is bypassed in the mixer's signal chain until we choose to activate it. Double click on (SC4) to bring up the plugin's GUI, and click on the red Bypass button – it should turn grey, to show that the plugin is now active. Start session playback with *Ardour*'s transport control, and the Amplitude meter in the SC4 plugin GUI should now be showing some activity.

The Gain Reduction meter probably won't be doing anything yet, because the default settings don't do any compression. You may need to bring the threshold level down to about -20dB or lower, and if the track has loud transients, you'll probably need to reduce the attack time to 10 milliseconds or so. The Attack control determines how long the compressor waits before applying the

effect, so if attack is set too long the plugin will miss transients, such as snare or bass drum hits.

If there are only a few loud transients reducing headroom across the track, a limiter might be the best solution. In the *swh-plugins* package there is a plugin called *Hard Limiter*, which, as the name suggests, will produce a harsh clipped sound if over-used. However, with more subtle settings it can be used for catching the occasional transient without audible side-effects.

To fix our third and final problem, of muddy and indistinct tracks, we can use a parametric equaliser, such as the LADSPA plugin *Triple Band Parametric with Shelves*. This EQ isn't like the one you find on some hi-fi systems, which is designed to compensate for acoustic deficiencies in the listening room. A parametric EQ enables you to specify any problematic frequency that you choose, and apply boost or cut as required. The bandwidth control enables you to set a wide boost or narrow cut across the frequency range, as desired. Knowing the exact frequencies to boost or cut is part of the mixer's art, but generally speaking, you're looking to avoid large overlaps in frequency bands between different tracks – and cuts usually work better than boosts. **LXF**
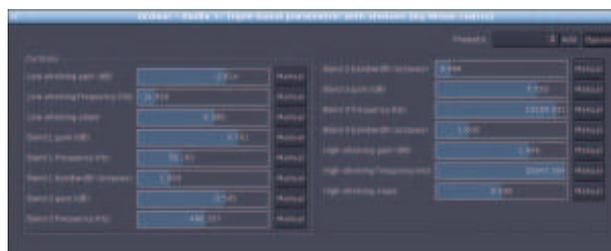
> "**Knowing the exact frequencies to cut or boost is part of the mixer's art.**"

> **This triple-band EQ with high and low shelves enables you to shape the frequency curve of your track.**

(deci)Bel Labs

Decibels are plotted on a logarithmic scale rather than a linear one (like degrees Kelvin), but there's not much maths required in practical use. A simple rule of thumb is that each -3dB drop represents roughly a halving in power, and +3dB gain represents about a doubling in power. Perceived loudness is a subjective measure that depends on several factors, but the *Ardour* meters will give you a good general idea of what's going on in the session, level-wise.

Don't confuse dBFS with dB(SPL), the measurement of sound pressure in air, which is often referred to by the general public simply as 'decibels'. For instance, if someone says 'the police shut down the gig because the meter was showing 130 decibels', they really mean a sound pressure level of 130 decibels, relative to the threshold of human hearing. Both measurements are expressed as a ratio, but they're measuring completely different things.

» **Next month** The series concludes, as we master our *Ardour* session with *Jamin*.