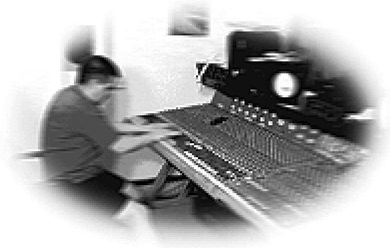
technique practical mixing

Practical Mixing

Hands-on advice to help you improve the fundamentals of your mixing technique.

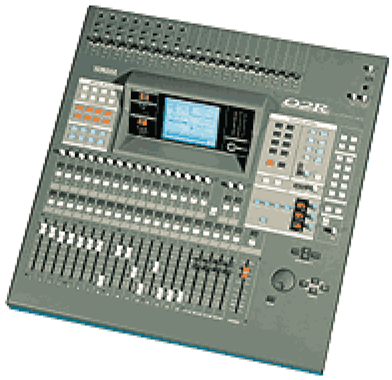


Hugh Robjohns

Whether you own a simple eight-into-two mixer, or a giant in-line multitracking console, the basic facilities and principles of their operation are the same, and knowing how to use a mixer properly can make the difference between a well-recorded master track and something which is distinctly substandard.

Starting From Scratch

To start at the beginning, always make sure the mixer is 'zeroed' before starting a new project. By that, I mean make sure the input gains are at their minimum settings, the polarity reversals and high-pass filters are switched off, the EQ is set flat and bypassed, the auxiliaries are turned down, and any group routing is deselected. In the case of a digital mixer you could store this as a default memory from which to start each new project.



With the mixer in this kind of 'safe' configuration you can start to build a mix coherently, without having to cope with unwanted signals turning up in all sorts of strange destinations. There is nothing so frustrating, unprofessional and time-consuming as having to hunt down the source of the strange noises coming from one of your effects units! At least with analogue mixers the offending knob can usually be spotted quite quickly, but with assignable digital consoles it can be a nightmare -- hence the importance of always starting from a known safe condition where everything is turned off. I have never understood why the purveyors of digital mixers seem to have a default condition with all the faders open at unity gain...

Gain Structure

The most critical aspect of mixing is getting your gain structure right. This means optimising levels through each part of the signal path so that the signals are kept well away from the noise floor at the bottom, and below the overload point at the top -- and it is especially important to allow sufficient headroom to allow peak transients through undistorted.

Something to bear in mind with budget analogue mixers is that headroom is often the limiting factor, especially when working with digital recorders. The relatively low voltage rails provided for the electronics mean that it is simply not possible to output signals with peaks in the +24dBu region often required to fully modulate a professional digital recorder. And, just because the mixer's meter is scaled to +10 dB and above, it doesn't mean it is able to drive clean signals that loud. It is worth experimenting with your particular mixer to see what it is truly capable of, and aligning the input sensitivity of connected equipment accordingly.

By way of a personal example, I have a Mackie 1402 VLZpro -- a very cost-effective little mixer which I use in a compact location recording rig -- which drives an Apogee PSX100 A-D converter. Although the specifications suggest otherwise, this desk is not capable of driving the peak levels required to fully modulate a professional digital recorder to +20dBu. However, by re-aligning the Apogee to read +12dBu as the maximum digital signal (0dBFS), the Mackie is never stressed to deliver peak levels and, as I only record direct to DAT and CD-R, the noise floor remains sufficiently low for 16-bit performance.

Another workaround I have used on many occasions to avoid stressing a budget mixer is to connect the nominally +4dBu professional-level output to the -10dBV input of semi-professional digital recorders. This effectively provides an additional 12dB of output gain without running out of steam in the mixer.

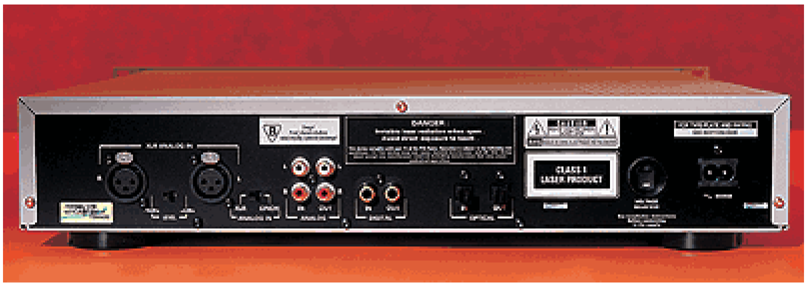
Important Monitoring Facilities

In most budget desks the monitor section is the weakest link, often providing only a level control, a headphone outlet, and external monitoring of a master recorder return. A professional desk will have a plethora of external monitoring sources, the ability to drive multiple loudspeaker systems in stereo or mono, and the provision to dim or cut the monitors without changing the setting of the level control.

This last feature is particularly important because, when mixing, your ears develop a level reference based on the listening volume. If you change the monitoring level, that reference will be destroyed, and your mix will be inconsistent as a result. Professional monitoring systems therefore provide Dim and Cut buttons to reduce the monitoring level or to turn the speakers off without affecting the original level setting. It is also important to be able to check the mono balance of a stereo mix, so professional consoles provide facilities to listen in mono (ideally through only one of the speakers).

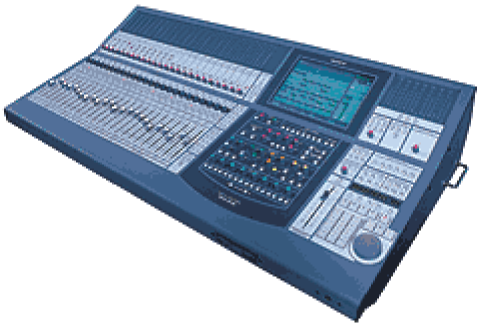
Mixing In The Digital Domain

The same fundamental gain structure considerations are also important in digital mixers, although the provision of sufficient headroom is usually far more important than worrying about the audibility of the noise floor. With an analogue system, as you run out of headroom, the amount of distortion buil ds relatively gradually -- some engineers deliberately run their mixers fairly hot to take advantage of the distortion (although few mixers sound nice when driven hard). However, digital mixers don't have headroom of their own -- you have to allow for it when you set levels -- and the moment you run out of headroom everything sounds extremely horrid.



Digital mixers come in two flavours: fixed point (most budget consoles) and floating point (usually the more expensive consoles). If you want to know more about these terms, check out the 'All About Digital' series which began in *SOS* May 1998. The important thing to realise is that floating-point designs have a massive internal dynamic range, which means that there is effectively infinite headroom and a totally inaudible noise floor within the mixer. Levels will still have to be managed to ensure the signal sits nicely within the output window of the digital or D-A interfaces, but you can't 'break' the signal while processing within the console.

On the other hand, a fixed-point console will have a finite dynamic range of about 6dB per bit of internal processing resolution. While this can look quite a lot on paper, it is surprisingly easy to run out of headroom if applying a lot of equalisation and mixing a lot of signals together. That is why these kinds of consoles tend to provide digital attenuation just prior to the EQ section. If you are planning to introduce substantial amounts of equalisation, or if a lot of signals are being combined, think about winding in a few decibels of digital attenuation first.



Getting Your Fader Levels Sorted Out

Whether you're using an analogue or digital mixer, it's designed to operate most effectively with its faders at or near the 0dB or 'unity gain' position (sometimes simply marked with a 'U'). This provides the best compromise between a useful amount of headroom and adequately low mix-buss noise. So, when setting up your mix, you should open each fader to the unity mark and then adjust that channel's input gain control so that the signal is at roughly the right level for the final mix. Clearly, you won't know what that is for sure, and EQ'ing the signal will affect its level too, but you can usually get it pretty close with a little thought and some experience.

If you have little experience with mixers, a handy tip is that when you have created something resembling the perfect mix with all the faders close to their unity marks, mute all the channels and then audition each one in turn to see how loud particular sources are in isolation. That way, you will have a good idea where to set the input gains for those instruments the next time.

If, when you come to mixing, you find all the faders are barely off the backstop, or pressed up hard at the top of their travels, you need to re-optimise the channel gains, because you are in danger of either running out of headroom in the channels or suffering unnecessarily noisy signals.

Optimising A-D Conversion

The conversion of analogue signals to digital data is something with which almost all recording musicians will have to contend. If you're using a digital mixer, analogue signals will need to be converted at the mixer's inputs. However, even if you use an analogue mixer, you're still likely to have to convert to digital at the inputs of your master recorder. As a result, it's important to know how to get the best out of the analogue-to-digital (A-D) conversion process -- a signal that is badly converted from the analogue domain to the digital domain will always sound bad, irrespective of what digital processing is available to you.

To get the best conversion, you should ideally set the input level to your A-D converters so that the peak analogue signal level is as close as possible to the maximum level accepted by the converter. However, there is often a degree of uncertainty about the maximum peak level of an analogue signal and so some headroom will have to be provided. This is particularly true in live situations where the signal sources are inherently unpredictable, but even in more controlled studio circumstances the most carefully optimised levels can fall foul of occasional mic popping, for example.

Creating this headroom solves the problem of transient distortion, but introduces two undesirable side effects. Firstly, for every 6dB of unused headroom you effectively lose one bit of digital resolution. With an unprocessed transient-rich percussion track you may well have to provide as much as 24dB of headroom to ensure those transients are captured intact, which means your 24-bit A-D converter is really only providing 20-bit resolution to the main body of the sound. With a decent delta-sigma converter there shouldn't be any problem with this, but with budget 16-bit converters the resulting quantisation noise and low resolution can prove a major headache! The second problem with leaving enough headroom is that noise levels can become a serious problem, particularly if digital gain and compression are applied to compensate for the low overall signal level.

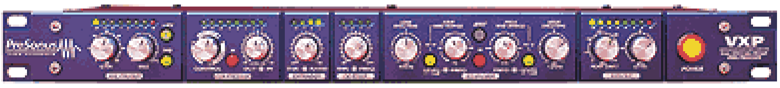
However, there are ways of dealing with these problems. Perhaps the most useful way of circumventing the need for excessive analogue headroom is to use an analogue 'voice-channel' processor to condition the signal -- at least partially -- prior to conversion. In this way the dynamic range can be brought under control and the peak-to-average level optimised more effectively before the signal reaches the A-D converter, providing a better sound overall and generally making life easier in the digital domain. The huge number of voice channels available on the market indicates just how popular this approach is at all levels of the music recording industry.

Input Conditioning

Most home setups rely entirely on nearfield monitoring, and the weak or missing bottom octave(s) of such systems can make it very hard to know exactly what is going on in the deep bass region. Vibrations through mic stands, plosive popping and air conditioning or other machinery can all create huge amounts of LF energy which, at best, clutters up the mix, and at worst makes it impossible to create a decent mix at all.

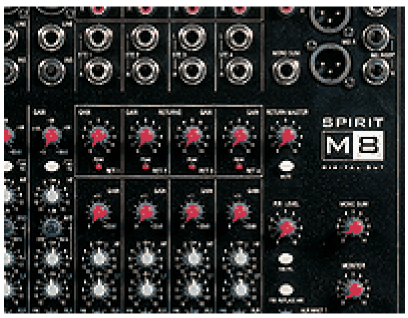
So if in doubt, filter it out! Many desks include switchable high-pass filtering on the mic inputs and some even allow a tuneable filter. After setting the input gain, I routinely flip the high-pass filter in to see if it improves the sound quality or not. At this point it is often also a good idea to apply some corrective EQ cut if it helps clean up any acoustic spill. For example, if you have a number of microphones in close proximity -- such as around a drum kit -- there will be a lot of spill between them. Just make sure that any EQ you're applying to sort out spill problems doesn't detract from the wanted signal unduly.

Another thing that can help with spill is judicious use of the polarity reversal switches that are provided on most mid-price consoles. However, this facility is mostly used to find the most suitable sound where any instrument has been captured using several recording methods simultaneously.



Setting Up Your Effects Sends

Even the simplest mixer usually provides one or more auxiliary outputs as well as the main programme output. You can use these to send signals to external effects units -- delay, reverb and modulation processors in particular -- while mixing down. Auxiliaries are always equipped with a level control, often a pre/post switch and sometimes an on/off switch too. The pre/post switch determines where the auxiliary signal is derived from in the channel signal path -- either before or after the channel fader. For mixing purposes, you'll probably want to switch these to post-fader, so that if you fade out the source it's effects processing disappears as well.

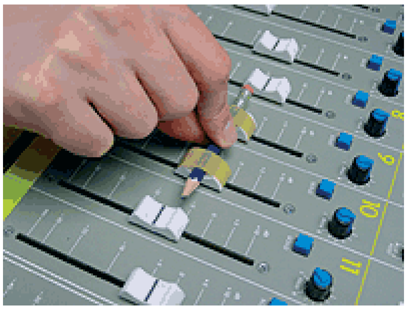


Obviously, if you're sending to an external effects processor, you'll also need to return the processed signal to your mix. Many desks have dedicated effects returns channels, but if yours doesn't have any (or not enough) you can always use any spare mixer channels instead.

Working In Stereo

A lot of sources are stereo these days, including drum machines, keyboards, turntables and so on. Ideally, you should bring these sources into the mixing console through a dedicated stereo channel -- most desks provide at least a few these days. The advantage of a stereo channel is that there is only one gain control, one fader to push and pull, and only one set of equaliser knobs to twiddle. If there are only mono channels in your desk, you will have to plug the left signal through channel one, say, and pan it hard to the left, while plugging the right signal through channel two, panning it hard right... and now you have two input gain controls, two faders and two sets of equalisation to worry about.

Why is this important? Well, the position of a sound in the stereo image is essentially governed by the relative loudness of the signal components in the two channels -- a pan control takes a mono signal and allocates part of it to each channel, the relative proportions producing the required image locat ion somewhere between fully left and fully right. In the case of a stereo signal, the stereo positioning of the instruments it contains is already defined -- that is why it's stereo. If you are running this through two mono channels and they are not perfectly matched in terms of their gain and equalisation, the imaging will be offset to one side or the other. In the case of a mismatched equaliser, the image will pull in different directions at different frequencies.



The only solution is to take great care in making sure the two channels are matched as closely as possible. Matching input gains is fairly straightforward, particularly if you can temporarily send a mono signal from the stereo source (use a mono record or CD, for example, or play a mono instrument sound without effects, rather than a stereo sound, from the keyboard).

When it comes to EQ, you just have to try to set the positions of both sets of EQ knobs to the same values and then hope for the best. Listen carefully to the stereo image and make sure you are happy w ith it once you have everything lined up as best you can.

Of course, once the gains and EQ are set, you probably won't want to mess with them again during the mix, but you will still be riding the faders up and down. This is where most people come unstuck, because if one fader is moved a smidgen more or less than the other, the stereo image will wander.

To help reduce the likelihood of an unstable stereo image, you need to couple the faders mechanically. Companies like Canford Audio and Studiospares in the UK can supply clear plastic clips which fit over the two knobs, effectively locking them together. These are supplied to fit over different-shaped fader knobs from different manufacturers, and span different widths, so you will

Working With Effects

The most common problem for anyone new to mixing is deciding how much reverberation to add. *Sound On Sound* has carried many articles in the past on using reverb -- particularly on selecting and fine-tuning the available programs of specific digital reverb machines. The idea of adding reverb is create the illusion of depth, adding a second dimension to the mix. In real life, the closer you are to the source, the stronger its direct sound will be in relation to the room reflections. The greater the room size, the longer it will take for those reflections to arrive, and the harder the surfaces, the denser the reverberation tail will be. Armed with these simple concepts, it is possible to modify the parameters of most reverb units to create believable acoustic spaces which will complement your music.

The thing always to bear in mind is that reverb will tend to clutter the mix by filling in the gaps between instruments, both in the time and frequency domains -- particularly if long decay times are used. Try to create the room sound you want just through the levels, density and timing of the early reflections, adding a sufficient decay to sound natural, but without it becoming intrusive. If you have more need to check out your specific requirements before ordering. A cheaper, though admittedly less attractive, option is to link the fader caps together using a short pencil and some ordinary sticky tape... Whichever solution you choose, make sure the link doesn't foul any routing buttons which may be positioned between adjacent faders.

than one reverb machine available, use different programs to create different acoustic dimensions and more interest in the mix.

In terms of balancing the reverb levels against the rest of the mix, the most useful advice I can give is to fade the reverb return up until it sounds about right, then back it off by about 4-5dB. This rule of thumb tends to cure the typical problem of having too much effect in the mix.

Starting Work

There are several different approaches to mixing, but most people like to start with the rhythm section, building the song up from there. Just to be contrary, I often work from the opposite direction, starting with the most important element of a pop song -- this will usually be the lead vocal track -- and then fill ing in the countermelody instruments and supporting sections as necessary. It is often easier to make sure the peak levels stay under control by working this way. Starting from the bottom and working upwards can often lead to a final mix which peaks significantly higher than intended.

Again, a popular approach is to start mixing with everything panned to the middle, and to start panning sounds to create a stereo stage only when the mix is roughly right. The drawback is that if you pan a sound close to either side, its level will fall by nearly 6dB relative to its level at the centre, which is enough to completely upset the balance. Consequently I find it quicker and easier to pan sounds to their appropriate positions in the stereo sound stage before attempting to build the balance.

It is important to keep checking the balance in mono as well as stereo (or surround) -- particularly if your material may end up on the radio. Inevitably the mono mix will sound different to the stereo (and the stereo to the surround), so a degree of compromise will be required to achieve the most acceptable results in each format. Pay particular attention to reverbs, which have a habit of drying up when you listen in mono.

Referencing Your Mix

It is very easy to lose your way when starting out in mixing. Your attention is continually being drawn in different directions, concentrating on the different instruments and voices. This makes it very easy to become bogged down in the detail of the mix, rather than its overall balance. You also have to consider the general tonal balance of the mix and how it compares to commercial tracks of the same genre.

One way to create and maintain a reference on which to base your mixing decisions is to listen to examples of similar music released commercially. Playing suitable music CDs through the same monitors in the same listening environment will help establish the all-important sonic reference with which you can judge your own mix. Listen critically, and form opinions about both the spectral balance and the relative levels of different instruments. You may also be able to come up with some good ideas for reverb and effects settings, and maybe even a few production ideas!

Another trap to be wary of is that of ever-escalating listening levels. In general, the louder your monitoring level, the less accurate your perception of balance will be. This is because it is easier to hear detail in quiet elements of the mix when the whole thing is very loud. Turn the volume down to a more normal domestic listening level and the quiet bits will be completely lost! Far better to set a comfortable listening level with your reference music sources (which may be a little louder than a domestic level, but not too much), and then try to balance your mix at the same level. It is a little harder to do, but you will get better, more transportable results, with less ear fatigue and happier neighbours...

At the mixdown stage, equalisation can be used to help instruments retain their individual clarity. I find that cutting unwanted frequencies generally sounds more natural and subtle than boosting wanted frequencies in this case. Whatever settings you choose, make regular use of the EQ bypass button to make sure you really are improving the situation and not just making everything sound loud and bright.

You Live, You Learn

The skills required to create a great mix don't come naturally to most musicians, and there are a lot of potential pitfalls for the beginner. However, if you take the time to practise the above mixing techniques, with a selection of reference CDs close at hand, you should be able to keep yourself heading in the right direction.

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

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