the big squeeze

COMPRESSORS & HOW TO USE THEM

These days a compressor is pretty much an essential purchase if you're recording vocals or non-keyboard instruments. PAUL WHITE looks at how they operate, and how they can be used both to control levels and to fatten sounds.

To the best of my knowledge, compressors were first developed as a means of keeping the levels of location movie sound under control, shortly after the industry decided that talking pictures had earning potential. They were soon adopted by the music recording industry as a means of keeping the vocal excesses of untrained pop singers under control, but along the way the benign side-effects of heavy compression became a production trademark. Indeed, compression is as much a part of modern music-making as digital reverb.

Though the use of compression is not always as well understood as it could be, the fundamental workings of these devices are pretty straightforward. Essentially, a compressor is a processor designed to reduce the dynamic range of an audio signal by applying gain reduction when the input exceeds a certain level. In other words, when the sound gets too loud, the compressor turns it down. In the context of pop music, this is a useful way of applying automatic level control to singers who may not be able to restrain themselves on louder notes. In addition, vocalists find some phrases and words easier to sing than others, and the outcome is usually a performance that fluctuates in level by a considerable margin from phrase to phrase -- and even from word to word. You may have experienced this in your own demos made without compression, where some sung words and phrases tend to be obtrusive while others get almost completely lost beneath the backing -- and, aside from the considerations of vocal intelligibility, unplanned changes in level make a recording uncomfortable to listen to. Furthermore, because pop music tends to have a fairly restricted dynamic range compared with, say, classical music, a degree of routine compression can make the vocal sit more comfortably at the correct level in a mix. Though vocals are the most obvious candidates for compression, most acoustic instruments work better in a pop context when their dynamic range is deliberately restricted. The same is true of electric guitars and basses.

One aspect of compression that causes confusion is whether it makes loud sounds quieter or quiet sounds louder. The mechanism of compression means that loud sounds are reduced in level, but most compressors have an output level control that allows any gain lost by compression to be restored or made up for. If you apply enough make-up gain to bring the signal peak levels back to where they were before compression, the quieter signals will be louder than before, so you can think of compression as both a way to make loud sounds quieter and to make quiet sounds louder. Figure 1 should help to explain this, as it shows an uncompressed signal, a compressed signal, then the same compressed signal brought up to the same peak level as the original. As quieter sounds can, in effect, be increased in level, compression has the effect of boosting the average signal level which, in turn, means that the average energy level is higher. This often results in a more powerful or punchy sound, even though the peak level is unchanged.

COMPRESSOR ACTION

A typical compressor comprises a gain control element, such as a VCA, and a photocell and diode arrangement, or an FET gain cell in series with the input signal. A second part of the circuit, known as the side-chain, monitors the input signal to establish its loudness or level. The signal level is continually compared with a threshold set by the user, and when the signal reaches or exceeds the threshold, a control signal is sent to the gain element to reduce the level of the signal. Though this might sound a little complicated in engineering terms, it's almost the exact equivalent of listening to a recorded track over monitors and pulling the fader down when you feel it's getting too loud. Indeed, manually controlling levels in the way I've just described is known as gain riding, and a compressor is simply an automatic gain rider. The problem with doing the job manually is that, unless you've played the track through and memorised exactly where the loud and quiet spots are, you'll always respond too late to changes in level, because you can't start to move the fader until you hear the start of the offending loud or quiet sound. Add to that the reaction time of a typical human being and you can see why you'll always be chasing the problem rather than curing it!

Before I explain how to set up a compressor, it probably makes sense to run through the various controls you're likely to encounter.

CONTROLS & TYPES

All the compressors I've ever used worked via a threshold system of one kind or another. With the simplest form of compressor, life is very black and white -- if the signal is below the threshold set by the user, nothing happens to it, but as soon as it reaches the threshold, it is turned down by a specific amount. In the case of what's known as a 'hard-knee' compressor, the threshold level is well defined, but in a so-called 'soft-knee' compressor, the gain reduction is introduced more gradually.

• **RATIO:** The ratio control is very important, because in most compressors this determines the severity of the gain reduction to be applied once the signal reaches the threshold. The higher the ratio, the more gain reduction is applied and the stronger the compression effect. If the ratio is made high enough, the signal level can, in effect, be prevented from ever getting past the threshold, and this situation is known as limiting. Though a limiter requires a theoretical compression ratio of infinity:1, any ratio above around 10 is so close to true limiting that it is usually referred to as such. Because most compressors have enough ratio range to allow them to be used as limiters, they are often termed compressor/limiters.

Ratio is defined as the number of dB by which the input level needs to increase to cause a corresponding 1dB rise in output level. If, for example, a compression ratio of 5:1 is set, an input signal exceeding the threshold by 5dB will cause only a 1dB increase in output level, as shown in Figure 2.

• **HARD KNEE:** As touched upon earlier, a conventional compressor has no effect on signals that are below the threshold, but as soon as they reach the threshold, gain reduction is applied at the ratio set by the user. This is known as hard-knee or hard-ratio compression because the onset of compression is sudden and occurs as soon as the threshold level is reached.

• **SOFT KNEE:** Because hard-knee compressor can sometimes sound a little abrupt or heavy-handed, the soft-knee compressor was developed. With this type of compression, gain reduction starts a few dBs below the threshold, but at a very low ratio. As the signal gets close to the threshold, the ratio increases, until at the threshold the ratio is that set by the user. Usually the ratio increases over a range of 10dB or so before the threshold is reached. This type of compression isn't quite as positive as hard-knee compression, but in some applications it can sound smoother and more musical. Figure 3 shows the characteristics of a soft-knee compressor.

Soft-knee compression is often used when the compression needs to be 'invisible', such as when you're keeping a mix level under control, whereas hard-knee compression is used in situations where it doesn't matter if you can hear the compressor working. Indeed, the audible side-effects of hard compression are often used as production devices to make vocals or specific instruments stand out in a mix.

TIME CONSTANTS

Earlier, I compared compression to the manual process of pulling a fader up and down. Just like the human engineer who does this, a compressor side-chain has a finite reaction time. It may be a lot faster than a human, but it's still true that a conventional compressor can't start to pull the signal level down until it has reached the threshold, and if that signal happens to be a snare drum with a near-instantaneous rise time, the compressor has to work incredibly fast to prevent the sound from shooting past the threshold level. In fact we don't always *want* to prevent the signal from overshooting as, in cases where brief peak overshoots aren't critical, the subjective result can actually be better than 'perfect' compression. For this reason, 'attack' and 'release' controls are provided to determine how quickly the gain is pulled down once the threshold is reached, and how long the gain takes to rise back to normal once the signal falls back below the threshold. Creating a deliberate overshoot by setting an attack time of several milliseconds is an effective way of emphasising the percussive nature of drums. Too short a release time can result in level 'pumping', while if the release time is too long quieter sounds following a loud beat may be reduced in level even further.

Setting the best attack and release values for a given type of material can take a certain amount of skill and experience, and if the programme material is constantly changing in dynamics, no one setting is going to be quite right -- which is why programme-dependent attack and release time were developed. An Auto function continually adapts the attack and release characteristics to the material being processed, by monitoring not only the input level but also the rise and fall times of signal peaks. Such systems can be very effective, especially on complex mixes or vocals.

If you were to set a very fast attack and a very fast release time, in addition to level pumping you might also end up with audible distortion, due to the fact that the compressor would be trying to work on individual cycles of the input signal rather than on its overall envelope. This phenomenon is particularly noticeable when the input signal is from a bass instrument, as the individual cycles are long enough to allow the compressor to respond. To get around the problem, it is necessary to increase either the compressor's release time or its hold time. Hold time is a short delay that prevents the compressor from going into its release cycle until a certain time has elapsed. All you need is a hold time longer than the wavelength of the lowest audio frequency and the problem is cured. Few compressors nowadays seem to include a variable hold control, but many have a fixed hold time built in, which ensures that the problem will never arise. If distortion does become audible at fast attack and release settings, and you don't have a hold control, you must increase the release time until the distortion stops.

SIDE-CHAIN SENSING

To continue comparing the compressor side-chain to the human hearing system... the compressor will always go by the average level of the sound rather than by the peak level, because the human hearing system tends to average out sounds in such a way that short, high-intensity peaks might actually sound less loud than a continuous sound at a lower level. That's one reason why the old-style VU level meter became so popular -- the sloppy response offered by VU meters is pretty similar to the way we humans perceive sound levels.

For a compressor to respond to averaged signal levels, it needs what is known as RMS level-detection circuitry. Such a system will invariably let short peaks slip by, and though this doesn't matter so much in the case of analogue recordings (where brief level excesses translate to brief increases in distortion) there are situations, such as when recording digitally, where peaks need to be better controlled. For that reason, some compressors are fitted with peak level detectors, which respond to signal peaks, no matter how short. In addition to keeping a better check on peak levels, these compressors can work better on drum sounds, where average signal levels bear very little relationship to what the signal is actually doing. Some compressors use RMS sensing, some use peak sensing, and some use a system that is somewhere between the two. Others give you the option to switch between one type and the other. Always try both settings if you're lucky enough to have a compressor that offers both. As a general rule, peak detection works best with percussive sounds.

STEREO LINKING

When you're compressing stereo signals it's necessary to ensure that both channels are subjected to exactly the same amount of gain reduction, otherwise the stereo image will drift from side to side whenever the signal in one channel is louder than that in the other. For example, if a loud sound occurs only in the left channel, the left channel level will be pulled back, and as a result the mix will appear to swing towards the right channel, where less gain reduction has been applied. The Stereo Link switch of a dual-channel compressor usually sums the side-chain inputs together, then controls both channels from the same side-chain. It may be necessary to set up both channel controls in the same way (the control settings are usually averaged, in this case), or you may find that one channel becomes inoperative and the other channel's controls affect both channels.

USING COMPRESSORS

A compressor should be patched into a mixer via an insert point, or connected in-line between one piece of equipment and another. Compressors should not normally be used with aux sends. It is common practice to add some compression to a signal while recording and then apply more at the mixing stage, should further control be necessary. This approach makes good use of the recording medium's dynamic range and, to some extent, protects against unexpected signal peaks. However, it's usually better to apply a conservative amount of compression during recording, which means you won't get as much protection against peaks as if you were hard limiting. Having said that, if you apply too much compression there's no easy way to undo it afterwards. Likewise, if the compressor has a built-in expander or gate, this might be better left switched off during recording, as a gate which has been set up badly can completely ruin an otherwise perfect take. Furthermore, if you save the gating until you mix, any noise inherent in the recording medium itself will also be gated out. If the gate settings are wrong, you simply reset the gate, then roll the recording again.

SIDE-EFFECTS

Perhaps the most common shortcoming of conventional compressors is the unwanted modulation of high-frequency sounds, due to large amounts of gain reduction brought on by high-intensity bass sounds. In most music, especially electrically assisted music, the majority of the sound energy emanates from the bass end of the spectrum, obvious examples being the kick drum, bass synth, and bass guitar. Any high-frequency sounds that occur at the same time as high-energy bass sounds will obviously be compressed along with the bass, and it's quite common to hear hi-hats and other bright sounds being pulled down unnecessarily. One way to get around this is to use a multi-band compressor that applies different amounts of gain reduction to different sections of the spectrum. In practice, though, these are costly and rarely sound natural. A more pragmatic solution is to set a slightly longer attack time, to allow the attack of the hi-hat, for example, to pass through the gain-control element before any gain reduction takes place. How successful this is depends very much on the design of the individual compressor and on how much gain reduction is being applied.

It's surprising how much the sound quality of different compressors differs depending on their design and on the type of gain-reduction elements used. Tube and FET compressors tend to introduce a little even harmonic distortion, which has the effect of brightening up the sound, whereas compressors based on photocells tend to sound quite gentle. Even VCA-based compressors can vary greatly -- unsophisticated designs often dull the sound or appear to cloud the mid and high-end detail, whereas a really good VCA compressor can sound almost perfectly transparent. The main artistic differences tend to occur when the compressor is being driven hard, which is why certain models are valued for the effects they create rather than for their integrity.

SUMMARY

There are almost as many different compressor characters as there are compressors, but there are a few basic rules that can be applied to setting them up.

• **LIMITING:** If you want to use a compressor as a limiter, mainly to control excessive peaks, you need to set the threshold fairly high and use a high ratio. The signal will then be unprocessed most of the time, but when a peak does occur, it will be controlled very firmly. A fast attack and release time is best, though if the sound appears to pump you'll need to lengthen the release time until the pumping is acceptable.

• **THICKENING:** There are times when you want to use a compressor just to thicken up a sound, and in this instance it's probably fair to say that you want to bring up the level of low sounds. To do this, set a much lower ratio -- perhaps as little as 2:1, or even less, but set the threshold quite low so that you still get between 6 and 12dB of gain reduction showing on the meters. A longer release time may give a smoother sound, but every sound is different, so let your ears decide.

• **SETTING UP:** Once you've decided whether you want to thicken or limit, setting up a compressor is quite easy and you don't really have to think about the threshold level much at all. Once you've set the ratio, adjust the threshold control so that around 6-12dB of gain reduction shows on peaks and you'll have a good starting point. You can then adjust for more gain reduction if you want audible pumping, or back off the threshold for less gain reduction if you want to be subtle. Always adjust the release time to be as short as possible without pumping -- start out at between a quarter and a half of a second -- and start with a fast attack. For percussive sounds, lengthen the attack while listening to the result -- you should set it just long enough to give the sound a good transient kick, and to avoid obvious gain modulation of high-frequency sounds. Smoother sounds such as vocals can be dealt with using a faster attack setting or, better still, an auto setting if you have one.

DUCKERS

You may know that compressors with side-chain inputs can be used to make one signal control the gain of another -- most of us are familiar with this technique through DJs using duckers to enable them to talk all the way through our favourite records. Personally, I prefer to use a gate with a ducking facility to create this effect, as I find it more predictable in operation and easier to set up, but you can use a compressor by feeding the signal you want to control into the main input, and the signal doing the controlling into the side-chain input. If, for example, music is fed into the main input and a DJ's voice is fed into the side-chain input, whenever the voice level exceeds the threshold, gain reduction occurs at the ratio set by the user. Figure 4 shows how ducking is achieved.

Ducking DJs (other than literally) isn't very inspiring, but you can use the effect quite creatively in a mixing situation by forcing parts of the backing track to drop in level to make a solo or vocal more audible. It's probably not a great idea to duck the whole backing track, but keyboard pad sounds or rhythm guitars could be usefully dropped in level by a dB or two for the sake of a clearer mix. If too much gain reduction is used, the gain pumping will become noticeable -- but many '60s hits pumped like mad, and they sounded great. Part of using effects is knowing how to make them sound good by abusing them creatively.

Ducking can also be used to control the level of effects such as delay or reverb -- indeed, many effects units now include the facility to do this automatically.

COMPRESSION AND NOISE

For every dB of compression applied, the signal-to-noise ratio is worsened by 1dB, assuming that the make-up gain is set so that the maximum levels of the compressed and uncompressed sounds are the same. This isn't because compressors are noisy, but because the quieter parts of the original signal, plus any noise it may contain, will be raised in level by compression. It is possible to use a gate to keep noise levels down, but care should be taken to minimise the noise at source first. If noise is a problem, it's essential to use as little compression (gain reduction) as you can get away with.

DE-ESSING

Some vocalists are more sibilant than others, and what starts off as a mildly irritating trait can become magnified out of all proportion by the time you've used your best capacitor mic, added a touch of 6kHz EQ boost for that extra sizzle, compressed the signal, and added a bright reverb. Fortunately, sibilance tends to occur in the 4-8kHz part of the spectrum, making it fairly easy to identify.

There are some units around which do an excellent job of reducing sibilance. These are known as de-essers, and the best ones act like compressors, but they only cut the section of the audio spectrum where sibilance occurs. However, it is quite possible to use a conventional compressor as a de-esser, providing it has a side-chain input and you have a spare equaliser. If the equaliser is patched into the side-chain signal path of a compressor, and set so that sibilant sounds are emphasised, the result is a compressor that responds more vigorously to sibilant sounds than to ordinary vocal frequencies. For example, if the equaliser is set to give around 10dB of boost only to sibilant frequencies, compression of sibilant sounds will occur 10dB before it does in the rest of the spectrum. Figure 5 shows how a compressor and equaliser may be used for de-essing. The shortcoming of this simple approach is that when sibilance is detected the level of the whole vocal is dropped, not just the level of the sibilant part of the spectrum. For this reason, you need to set a fairly fast attack and release time for the compressor, and settle for only a moderate amount of improvement, otherwise the voice will sound 'lispy' every time the compressor operates.

SOUND ON SOUND

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