An Engineer's Guide to Compression and Limiting

By Denis Degher

Specific examples of compressor and limiter settings can provide a valuable starting point when using them to process instrument and vocal sounds.

Compressors and limiters have become to the recording engineer what lens filters are to a camera operator: a way of controlling dissident sound or light elements for enhanced spectrum placement.

You can use compressors and limiters to increase the overall level of program material and hence improve the signal-to-noise ratio, and as a creative device for processing sounds. Compression and limiting ratios, attack and release times, and the degree of compression can help define an engineer's trademark sound, and many of us use specific hardware to record specific instruments and vocals.

The positive implication of this approach is that there is *no* substitute for experience. The negative implication in the artistic world is that mistakes play a major role in the creation of unique techniques and sounds. As with any creative tool, it is the *user* who determines the application and execution of a processing device.

The main control parameters of compression and limiting are peak and averaging (RMS); their response signal levels more or less parallel that of peak and VU meters. An averaging device responds to a signal's average level, like the human ear and a VU meter, and therefore tracks the overall power rather than the transient peaks.

The result is a smoother response characteristic. A peak-responding limiter, on the other hand, is designed to track the audio program peaks like a PPM; it is therefore much quicker responding and can catch instantaneous spikes. Because many of today's most popular compressor/limiters do not offer variable attack and release times, different devices are often used to suit the nature of the sound to be controlled.

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Sound examples

By compressing a **bass drum** by 1dB to 3dB or more during mixdown, using a compression ratio between 2:1 and 4:1, that tight "popping" or "clicking" sound heard on many dance singles can be achieved, without using an inordinate amount of high-end EQ. By using an averaging compressor with a slow attack time, the first transients of the signal envelope will be allowed to punch through before the compressor reduces the sound level above threshold. The result is a sound with a lot of "attack," but without the excessive amount of level that might result from a highly equalized bass drum.

Some form of compression is usually necessary to even out VU levels between notes.

A variation on this theme is to mult the bass drum. By keeping the primary-kick signal natural-sounding for thump, and compressing the paralleled kick signal for click, you can blend the two for power *and* click. By using a very fast attack time the leading edge of the sound envelope can be compressed as well, creating a unique but powerless bass drum effect.

Snare drums can be processed similarly. By heavily compressing the multed snare with a slow release time, a certain amount of sustain can be added by the pumping action of the compressor release. Radical equalization of the compressed channel can be added to create a highly processed sound.

Tambourines and other percussive, metallic sounds that have very high transient peaks can often cause circuit overload and tape saturation. In order to prevent this, peak limiting with a very fast attack time and a compression ratio between 8:1 and 20:1 can be used to help tame those excessive peaks on the leading edge of the sound envelope.

Percussive instruments that are played with sticks or various mallets of wood or metal (including marimba, vibes, steel drum, glockenspiel, tubular bells, triangle, cowbell and timpani) often produce intense overtones and transients. In fact, the overtones of certain notes may almost be as loud as the primary frequency, creating vast swings in level between notes.

Because of the cutting nature of high-frequency instruments, where they will be placed in the mix and the fact that they cause high crosstalk levels between adjacent tracks on analog recorders, correct mixing technique and lower recording levels without limiting or compression is usually more effective than peak limiting or compression.

Midrange and lower frequency instruments are a different proposition. They do not cut through a mix nearly as well as higher frequency instruments, and still may retain heavy harmonic overtones. As a result, some form of compression with relatively fast attack and moderate release times is usually necessary to even out VU levels between notes.

Low-frequency instruments, such as **electric bass**, often need leveling for a variety of reasons, including the way in which the pickups are set up, the pickup's frequency response or possibly because of the way the musician is playing. Certain notes tend to jump out audibly and need controlling. Another problem can be caused by level discrepancy between certain notes, which will be obvious when viewing levels on a VU meter.

To ensure proper tape levels, compress the sound by up to 4dB or 5dB, with a 2:1 or 4:1 ratio. If the attack time of the compressor is set too fast, the initial envelope of the note can be affected adversely, causing a punchless, "rubbery" sound. Another problem may occur while using fast release times on low-frequency sounds. Reducing the release time should ensure that the compressor threshold does not pump between the long, low-frequency cycles, or unduly affect the end of the sound envelope.

When recording an **acoustic bass,** many of the above problems are exacerbated by the instrument's acoustical characteristics and the recording environment. **Synthesizer bass** can also create significant level problems, some of which can be reduced by using low compression ratios. For shorter, sequenced **bass synth** sounds, a large amount of compression followed by heavy expansion at a high ratio forces the sound envelope to push its way through the noise gate, thereby creating a tight, expanding sound.

Synthesizers seem to need a certain amount of compression, because no longer are you dealing with a *natural* sound that must be treated carefully to retain certain inherent sonic qualities. In many cases, radical processing may help make the sound more *real.* **Synthesized horns** parts, for example, can often be greatly enhanced by RMS compression at ratios between 1.5:1 and 4:1, and compression from 5dB to 8dB or more, depending on the characteristics of the patch and compressor being used. A thicker, punchier and more dense sound can be achieved as the compressor positively affects the synthesis overtones, by squashing them together.

Synthesizers need a certain amount of compression because you are no longer dealing with a "natural" sound.

Percussive synthesizer sounds often contain jagged and irregular peaks that can be controlled various ways. For level control, use higher limiting ratios (8:1 or greater) on a peak device, or the peak setting on switchable unit, to control excessive level swings without affecting the lower-level program material.

Another approach is to compress the sound at low ratios to reduce both high- and low-level program levels. Again, this technique depends on the sound being created, because you are dealing with synthetic sounds. With **samplers** and **sampling keyboards**, particularly 16-bit devices, the question of what's *real* may become a moot point when sound quality is compared to the analog recording process. The latter's inherent weaknesses and compensations include pre-emphasis, de-emphasis, NAB and AES equalization, record- and playback-head gap loss, and so on.

So, when dealing with high-end sampling technology, some caution must be used to retain the "natural" sound, whatever that may mean today.

The **electric guitar** seems to enjoy compression for many effects. When close-micing a loud rock-and-roll guitar amp, squashing the sound at low compression ratios with a slow release time adds sustain, and definitely makes it easier to place in a mix. Another trick is to heavily compress the ambient mics by between 6dB and 10dB, thereby creating a denser, sustained ambience.

Although heavy compression makes it easier to place a raucous guitar in a com-

What is Gain Reduction?

There are many misnomers and misunderstandings about how compressors and limiters function. In an attempt to shed some light on their mode of operation and the different types of gain reduction control, RE/P offers the following definitions:

- *Leveling is usually taken to mean the long-term control of signal levels. The compression ratio is greater than 10:1, with slow attack and release times. As a result, leveling has negligible or no effect on short-term changes in the average level, or on sudden transients.*
- *Compression is used to maintain a constant power level; in essence, it prevents levels from falling too low. Benefits include improved signal-to-noise ratio and correction of excessive volume level.*
- *Limiting prevents a signal from exceeding a preset level and provides protection from overmodulation, overcutting and monitor damage, and elimination of transient peak levels.*

There are two basic types: program limiters (referring to the highest VU level occurring within source material); and peak limiters (regarding waveform peak amplitude, measured with a peak program meter or oscilloscope).

When the waveform amplitude cannot be allowed to exceed a predetermined value, peak limiters are used. They are characterized by fast attack and release times, high compression ratios and high threshold levels.

The parameters that characterize limiters and compressors are as follows:

- *Compression ratio defines an input-level decibel change corresponding to a user-defined, output-level decibel change. Perfect linear circuits produce a 1:1 compression ratio; compressors and limiters use ratios or slopes that range from 1.1:1 to 30:1. Variable-slope compressors begin with a low compression ratio, and then increase as the level of the input signal rises.*
- *Threshold refers to user-definable, dynamic range modification. All limiters and most compressors operate below this threshold level as linear amplifiers, with a relative slope of 1:1. In order to be effective, limiters usually operate at a threshold above 0VU.*
- *Attack time is the time required by a device to bring an input signal under 90% control after it has exceeded a defined threshold. To prevent sudden signal increases from escaping amplitude control, limiters usually incorporate fast attack times. To prevent unwanted effects on transients, compressors use slower attack times.*
- *Release time is the time required by a device to restore an input signal to 90% of full gain, after it has dropped below a defined threshold. While longer release times result in less apparent volume, faster release times can provide greater volume, but can also introduce more distortion into low-frequency program material.*

plex track, it can have the negative effect of making the overall sound appear smaller. As a result, compression should be approached with care, or avoided, when recording a power-rock trio where the guitar is the only chordal instrument.

When recording direct or close-miced **chordal guitars**, high compression with correct equalization can create a brilliant, shimmering sound with little or no attack. The sound appears to swell after the strum because of the pumping action of the compressor's release.

To help keep **lead guitar solos** highly audible, to add sustain and to avoid level problems during mixdown, compression may be used to compensate for vast level swings between guitar notes. Such level problems may be caused by a variety of factors, including the guitar's fretboard design; the tubes in the guitar amp, which may feature different harmonic frequencies; or the musician's pick attack during the solo, which may send differing amounts of voltage from the pickup.

For **R&B-style, single-note picking**, a slow attack time will allow through the percussive attack before compression begins, and a quick release will pop the level back up between notes, creating the "percolator" effect.

When played as a solo instrument, the **acoustic guitar** usually requires no limiting or compression. Unfortunately, in order to fit a plucked or picked steel string into today's complex mixes, something has to give, and it's usually the guitar's natural fidelity. This is not always bad; many of today's most popular steel string sounds are highly compressed. An example is the popular rock/pop strummed sound that features the string sound over the instrument's resonance.

The **gut string guitar** is another instrument that sounds much better in its natural form than when it's been compressed or limited. For placement in a pop mix, however, it is usually necessary to reduce the dynamic range. Experimentation is the only solution.

Because of an **acoustic piano's** highly transient, percussive nature, level differences between notes and the great dynamic-level possibilities, many engineers favor mild compression with a slow attack time. This lets through the percussive leading edge of the sound envelope before the compressor kicks in.

Personally, I prefer a natural sound with no compression and take my chances with tape compression of the high transients, unless I'm going for that highly compressed, altered sound heard on some records. That particular effect can be achieved with a medium-to-quick attack time at low ratios plus heavy compression, and depends on whether you want compressed sustain (a slower release) or the end of the note to pop back up (a fast release).

Occasionally, real **horn sections** will need a decibel or two of compression at

Excessive compression can bring up the breath factor too much in comparison to the note.

low ratios to blend them onto one track, or in stereo to maintain an even balance between instruments. Overcompression may result in an unnatural sound. **Horn solos** may also need touches of compression to avoid tape saturation on peaks.

Flutes of all types can produce large level jumps between notes that wreak havoc on RMS meters, but without being aurally noticeable. Another problem with flute is the breath component of the overall sound—excessive compression for note leveling can have the adverse effect of bringing up the breath factor too much in comparison to the note. The problem should first be tackled with mic placement; finding the correct mic(s) and proper orientation to the flute's mouth piece will be far more successful than using heavy compression or limiting.

Some engineers like to compress the close mics on **violins** and **violas** for blend control. I refrain from compressor limiting of strings, and prefer an ambient, open-micing technique that uses the room acoustics to capture the sound. The only exception to this rule are **cellos**, which usually are more closely miced and sometimes require slight compression to even out different notes caused by terse bowing, or from a pizzicato section.

Theoretically, no compression is the best compression when recording **vocals**. However, this cannot always be accomplished on many of today's dense, rhythm-oriented pop records. Finding the correct vocal perspective within the mix is possibly the most critical aspect of mixing such tracks. The artist must be readily audible without totally dominating the track. Recording vocals requires both sensitivity in dealing with the artist and attention to sonic integrity.

Many considerations enter into the recording of vocals, including the type of microphone being used, the room sound, the use of compression, what device is used and the vocal perspective in the cue mix, which in turn will affect how loud the vocalist sings. Just as certain microphones flatter certain vocalists, certain compressors may also prove to be complimentary to specific voices, depending on the unit's ballistics. What makes certain vocalists a challenge to record are large swings in dynamic range, in some cases 20dB or more between a song's pianissimo and fortissimo sections.

What further complicates the situation is that some voices do not readily accept compression without compromise, resulting in an unnatural sound. For voices that do not like compression, use very low compression ratios, light compression, moderate attack and release times, and gain riding to retain a more natural sound.

By feeding the compressor from the mic pre-amp, very little effective control is provided, and large amounts of level may reach the compressor and cause overcompression. By connecting the compressor across the console's channel bus, the input channel fader can now be used to control the amount of gain that reaches the compressor, thereby avoiding overcompression.

Certain artists with thinner, less intense voices crave compression, and may actually benefit from 5dB to 7dB of compression at ratios from 4:1 to 2:1. The use of compression seems to positively affect the harmonic overtones, creating a richer, thicker and more powerful sound.

Background vocal ensembles may be treated in a variety of ways, depending on the desired sound. If the group cannot blend with itself from a level perspective, compression for level control may be necessary. If you are after punchy, grabbing backing vocals, heavy compression can be used for effect. Another trick that some engineers employ is level *ducking*, where certain instruments are keyed to drop in level when the vocal enters, allowing for placement within the mix, rather than the "over-the-top" perspective.

Heavy compression (8dB to 20dB) of **ambient micing** represents another effective use of compressors that seems to be in vogue for many of today's records. This effect can be accomplished by compressing overall room mics, or keyed ambient mics.

Another area for creative compression is **special effects,** including reverbs, echo and delay lines. Depending on whether you are compressing the sends or returns to such outboard units, certain parts of the sound can be affected, depending on the device and the effect you're after.

Compressing the **stereo bus** during mixdown before the 2-track is a popular effect, particularly on technique consoles that feature an onboard stereo program compressor. This technique can smooth out the entire mix, but may prove unnecessary when various tracks requiring compression within the mix have already been processed. (The net result of this trick is to simulate the heavy compression that occurs prior to the signal reaching an AM or FM transmitter, thereby giving a single or album mix the same sound whether it is played over the air or on a home stereo system. Such processing has serious implications, however, since it can take the life out of the mix by inhibiting the song's dynamics.)

The advent of Compact Disc and digital multitracks in the studio is changing the way we approach recording, and the way in which compressors are being used. With digital's greater dynamic range and absence of tape hiss, packing the tape for maximum level is no longer a major priority. Because the 16-bit binary code does not recognize tape noise like analog tape, lower levels can be recorded without sacrificing signal-to-noise gains.

Care and taste must be exercised when using these compression techniques—they do not apply to all sounds.

In the digital domain, however, a new reason for compression or peak limiting exists, because digital recorders are totally unforgiving when the bit ceiling has been reached—unmistakable distortion and clipping will result. Because the digital domain does offer greater dynamic range, mixdown compression may still be desirable for sound placement within the stereo mix. Using compressors for creative sound, processing will be appropriate for achieving effects that have become popular.

As a recording engineer, I have found that care and taste must be exercised when using these compression techniques, simply because they do not apply to *all* musical styles and sounds. Compressor/limiters can have a great positive value, but may also have a negative impact on audio fidelity, transparency and transient response, making some material sound too clean, controlled and static.

Compression should be used appropriately to achieve specific sound goals, not as matter-of-fact processing.

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Acknowledgement: Thanks to Joe Chicarelli, David Sherfey, Peter Chaiken and all the other engineers in the recording industry who may have invented many of the tricks we take for granted.