Compressors and limiters: their uses and abuses

Mike Beville*

*Audio and Design Recording

The compressor-limiter is an extremely valuable tool in the field of creative music recording. Like any device, however, the greater the understanding an engineer has of its operation, the better will be the results obtained from its application.

TYPICAL PURPOSES of limiting and compression are to provide system overload protection, so avoiding distortion or even damage; reduce dynamics to make a more acceptable or comfortable range of sound level; and increase loudness or create impact.

Function

The compressor-limiter is primarily a linear audio amplifier with a voltage-controlled attenuating element. The control voltage is derived from the signal being processed in what is known as the 'side-chain'. The characteristics of the side-chain will determine the dynamic performance of the system: its sensitivity will establish the threshold level (the point at which gain reduction commences); its loop gain above threshold will control the slope or ratio (the relationship of input to output level); the way in which it integrates and derives the control signal will establish its attack characteristics, whether it be peak sensing, averaging or rms. The speed of operation, or attack-time, will depend on the integration time and any additional CR network. The recovery or release-time is normally controlled by CR networks, either singly or in a multiple arrangement. It is primarily the range of attack, release and slope characteristics which will determine performance possibilities and application.

Limiting

Limiting implies the use of a level control system to give overload protection; its purpose is to 'limit' the signal at a specified level. The amount of overshoot (the amplitude of a transient allowed to exceed the steady-state limit threshold) will be determined by the attack time. The effect of stopping every transient, no matter how fast, is likely to result in a lower average level, with audible side-effects such as a 'gritty' sound and switching spikes. Delay-line techniques, of course, can have a zero overshoot without these side-effects, but will still result in a lower than average modulation level.

There is a growing consensus of opinion which suggests that it is better to have limiter attack-times of some 250 μs to 1 ms, allowing the very fastest transients (which will not be visible on a ppm) to overshoot and in the extreme instance saturate the tape. Such an approach preserves the wave-front information that essentially gives the transient its characteristic, reduces side-effects within the system, and increases mean level for a given amount of gain change.

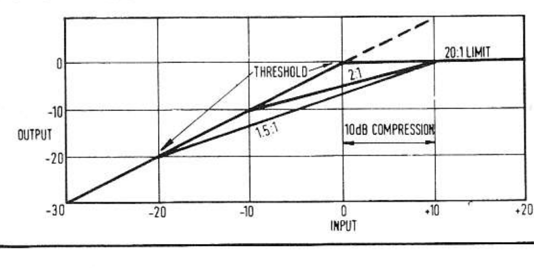
Where ultra-sensitive systems are involved—as in the case of am transmitters or pcm links—a diode-clipper is usually incorporated. Extensive tests conducted by the BBC several years ago demonstrated that a limiter with a medium attack time followed by a diode-clipper some 2dB above the limiter threshold, sounded more satisfactory than using a super-fast attack time.

The compression ratio in a limiter will need to be greater than 10:1, and will typically be of the order of 20 or 30:1 (see **fig. 1**). Although units are available with even higher ratios (100:1), it will be appreciated that in normal use the difference between 20 and 100:1 will be microscopic in terms of increased output, and the tighter slope will be certainly more audible.

The action of limiting must involve a peak-sensing side chain as it is peak level that is being controlled. When limiting, programme dynamics are not greatly modified since gain reduction—when it does occur—is usually momentary, of small magnitude and relatively short duration (a fast release-time being usual so that the action of recovery is inaudible).

However, 6 dB of limiting can make all the difference between background noise being audible or inaudible. The action of limiting thus allows an engineer to reduce his 'headroom' or overload margin, and thereby extend the dynamic range of his recording or transmission medium without fear of overload. In the limit mode, the compression ratio is said to be 'tight' because whatever the increase at the input, the signal level at the output cannot rise significantly.

FIG.1 COMPRESSION SLOPES



Compression

Compression is used to describe conditions of gain reduction that are more or less continuous; the original dynamics are compressed or reduced. Compression ratios may be anything from the softest slopes (typically 1.5 or 2:1) to the tightest 'limit' slope, dependent on the effect required. The ratio simply specifies the relationship between the input and the output levels. The normal relationship in an amplifier is 1:1; in a compressor or limiter this relationship changes above the threshold point and the output level rises at a slower rate compared to the input. For example, if a 2:1 slope is selected, for every 10 dB rise above threshold at the input the output increases by only 5 dB (see **fig. 1**).

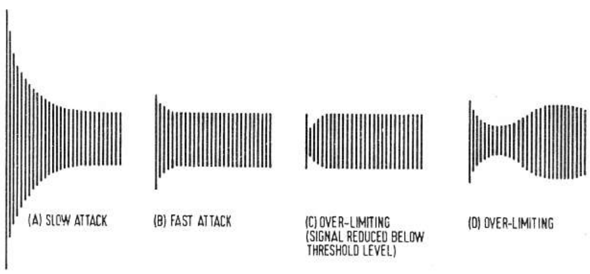
When it is desirable to compress, yet retain the maximum dynamics within the signal content, it is preferable to use a soft slope with a slowish or multiple release time. For a given amount of compression or gain reduction, the threshold on a soft slope will be lower than for a tight limit ratio. The same compression effected on the two slopes will sound different: on the 2:1 slope it will be hardly detectable, while at 20:1 it will be more noticeably stopped or limited.

Attack time

Attack time will determine the characteristic and size of peaks

 PRINTED IN ENGLAND

FIG. 2 ATTACK CHARACTERISTICS



allowed to pass through the system prior to attenuation; in effect it will dynamically modify the static sinewave response of the compression ratio. Slow attack can be observed on a ppm as overshoot, and is apparent aurally as a softening or easing on a tight limit ratio. As attack time lengthens, a subtle change takes place in the spectral energy balance as increasingly high frequency content passes unattenuated and, in extreme cases, can lead to sibilant accentuation. Slower attack times are useful especially when considerable compression is required with a tight ratio for maximum impact on an instrumental track (eg bass or drums).

Fig. 2 shows various attack characteristics on a pulsed sinewave; **2a** and **2b** show good waveform envelopes as the signal is smoothly attenuated to the threshold level; **2c** and **2d** are examples of over-limiting and poor design, and will sound constricted.

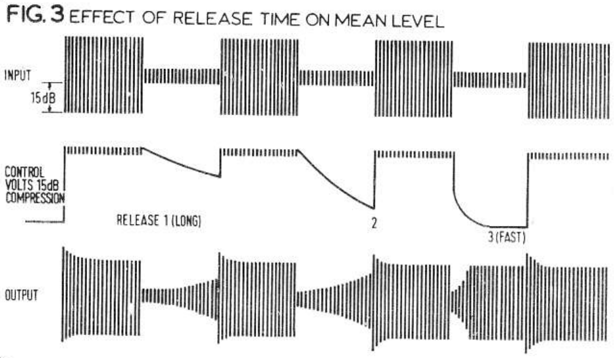
Release time

Release time is very important since it determines the moment-to-moment gain change in the system, which in turn controls loudness. Under conditions of considerable compression, very fast release time and tight ratio, the medium and low level signal content is raised to peak level (see **fig. 3**), thus increasing *subjective* loudness. (The definition of subjective loudness: sounds louder, but is at the same peak level). In the extreme, fast gain change becomes noticeable as 'pumping' or 'breathing'—the effect of background ambience and reverberation rising and falling in level. Pumping and breathing can be used for effect, but when unwanted can be minimised by either increasing the release time (or using a programme-controlled release network), reducing the amount of compression, or softening the ratio.

A programme-controlled release is obtained by means of a multiple network that gives two or more release times, dependent on signal level. It is intended to provide maximum gain change without pumping effect. Usually this means a fast release over 4–6 dB gain reduction before turning into a medium or long recovery time. The effect is sometimes described as a gain riding platform, and is ideal when considerable overall long-term compression is required (eg am broadcasting).

Where a fast rate of compression is essential, side-effects can be greatly reduced by recording in a dead acoustic with good separation and compressing prior to tape. By reducing reverberation, ambience and any cross-mic pickup there is little to indicate that gain change is occurring, and the engineer may be surprised at how much compression is possible. It should be noted that as the release time becomes shorter, low frequencies are increasingly flattened by the attacking action on each cycle. Fortunately the ear is very tolerant of lf distortion. In practice, therefore, this is not a major problem and can be used for effect (slowing the attack 'rounds' the distortion), and in any case is completely under the control of the operator. For bass instruments a release time of greater than 0.4s will give a totally clean sound.

FIG. 3 EFFECT OF RELEASE TIME ON MEAN LEVEL



Noise and modulation effects

Self-generated noise in compressors is rarely a problem in professional units. However, source noise can be raised through the action of compression on acoustic noise such as ambience, rumble, tape noise and spill-over from other instruments. If you compress off-tape, a 15dB gain reduction means an increase of 15 dB in tape noise (unless you use an expander). Even so, one should remember there is little or no masking of hf noise with a bass instrument, and it will be best to obtain the required sound before going onto tape.

Modulation of the signal by specific instruments can best be avoided by compressing individual or groups of similar instruments. There is then no dominant line to modulate another. It is impossible to limit a high-level lf signal without a most obvious and objectionable modulation of high-frequency signal and ambience (unless bandsplitting techniques are used). Such sounds must be treated as a separate track—get it tight and well controlled on or before the final mix-down.

There is often the need to compress a balanced programme where the dynamic range of the new medium may be more restricted. Modulation effects can be minimised by using a soft slope, a programme controlled release or an averaging side-chain. Some units incorporate or make provision for the insertion of equalisers into the side-chain to modify system response. Cutting the lf content will reduce any modulation effects caused by bass instruments, so that compression is controlled from the mid-band signal. This can only apply to compression, since limiting may produce sudden surprises as lf signals exceed the established limit threshold.

Boosting frequencies in the side-chain can also be used to advantage. An hf lift can predispose the compressor to operate on sibilants—with a variable frequency equaliser the engineer can find and boost the sibilant frequencies. Normally a tight slope would be used, along with a fast release and attack time. Compressor gain would be adjusted so that attenuation only occurred in the presence of sibilant signal. This is best done on a separate vocal track to avoid modulation of the whole programme.

Compressors need to incorporate some system gain (typically 20–30 dB), which means that normal line levels can be compressed by the amount of gain available, yet still appear at the output at the standard operating level in the chain. This allows comparison between the direct and processed signal.

A conventional compressor-limiter usually offers a range of ratios (eg 2, 3, 5, 10 and 25:1), but while the operator might prefer to use the softest slope (2:1) this can only be done on a well controlled signal. On a more unpredictable signal (for example, vocals), one might feel the need for overload protection as well as compression. On such a system a compromise must be struck by selecting a 5 or 10:1 ratio, which may not be quite so good artistically. In the more exotic units, it is often possible to compress at any ratio yet retain a limiter slope over the compressor. It is usually possible to vary the relationship between the compressor and limiter thresholds, so determining the amount of compression before the onset of limiting. Simpler systems may adopt fixed thresholds which, after say 10 dB compression, tighten to the slope of a limiter. Both approaches would allow the use of the softest slopes without fear of overload.

Expansion

The addition of an expander or gate greatly adds to the effectiveness of a unit. Besides reducing the increased source noise due to compression, the expander section can clean up tracks and dramatically reduce cross-mic pickup from other instruments. Gates are rather like limiters in reverse: typically, for a change of 1 dB at the input, the output falls by 20 dB, the rate being dependent on release setting. They can work well on punchy, well-defined dry sounds, but due to the switching type characteristic are usually critical to set up.

30 ▶

COMPRESSORS AND LIMITERS: THEIR USES AND ABUSES

The expander may be likened to the compressor, except that it is operating on a low-level instead of a high-level signal. The softer the slope, the easier it is to use without modulation side-effects; but the softer the slope the less useful it is in attenuating noise effectively. In most recording applications, the purpose of an expander gate is not to expand the music, but to get in below the low-level signal and attenuate the channel gain in the presence of noise only.

Imagine that on a particular programme the noise lies 10 dB below the wanted signal. By setting the expander threshold just under the music, it will be possible to lower the noise by a further 10 dB with a 2:1 expand slope. At this point it will be held on the noise itself. A tighter slope, of say 4:1, would increase the separation to 40 dB but the more susceptible to modulation effects.

For general track attenuation, a combination of peak sensing and averaging appears to be the correct side-chain characteristic, so that it opens quickly when necessary yet modulation effects are reduced on decaying signal. For creative work the peak sensing side-chain may offer greater options.

Matching

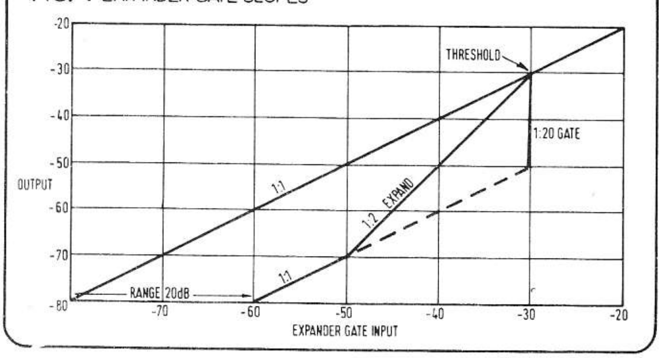
Stereo matching is a very important aspect since gain reduction must track very closely if there is to be no image shift during compression. Although mono units are sold with coupling possibilities, the potential user should establish the manufacturers stereo matching tolerances, since this is rarely stated. Purpose-built stereo units are likely to be more predictable in performance.

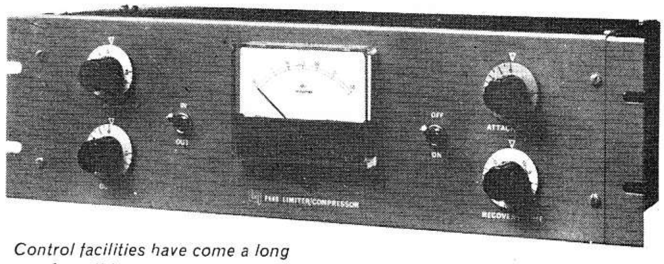
Metering

Gain reduction or compression is normally indicated on a meter or light system. Whatever the quiescent gain established in the device, this shows the moment-to-moment gain reduction occurring in the channel. In a combination unit, it could be limiting, compression or even expansion (although in the latter case there is usually some other indication of what section is functioning). In addition to showing gain reduction, it is customary for the meter or light column to give a visual indication of release-time. Clearly in the case of the meter, accuracy depends a lot on the ballistics, but it is normally a good guide to the rate of gain-change going on in the system. One should remember that although there may be a high compression reading, this does not necessarily mean that there is a lot of gain change operating—an indication of increased loudness will be the *rate* at which the meter moves. For little movement there will be little increase in loudness (other than the long term increase in lower level signal); where the change in dynamics within the music is faster than the release time, the music will hardly be affected whatever the amount of compression shown. It may be said that *the rate of gain change determines loudness.*

Vu metering is widely used in studios. However, since it doesn't indicate peak level a variation in setup procedure may be considered advantageous when using a peak-sensing compressor-limiter. If it is set up on tone, the vu will probably under-read by 6 dB when operational on a compressed or limited dynamic signal, and only approach zero vu under conditions of fast gain change. Bearing in mind that any system that uses vu monitoring must have good headroom (peaks in excess of 10 dB being not uncommon), it makes

FIG. 4 EXPANDER GATE SLOPES





Control facilities have come a long way from this: see survey p 34.

sense to set up under dynamic conditions so that the vu meter reads zero vu at least. Using fast attack in this circumstance the engineer can be confident that peak level is being well controlled some 6 dB higher without fear of sudden overload—well within the normal operating range. This may not apply to an rms or averaging device where peak levels may be less predictable.

Application notes

In any recording work, whether classical or pop, it will be best to apply compression to the sections needing it, rather than overall. Where this is not possible, gain reduction will probably be restricted to some 6–10 dB, if its effect is to be inaudible. Up to about 6 dB can be accomplished as limiting with a fast release (fast enough so that recovery is inaudible); over this it may be best to use an automatic release network, where possible, with a soft slope and the limiter coming in on top. In this way maximum dynamics are retained.

Studio

The effect of compression on signals containing plenty of presence frequencies, especially with ambience (ie choral work), is for the signal to recede as gain reduction takes place. Using the soft slopes will allow the sound to really get louder and reduce the impression of a receding image.

Using a tight slope on bass or bass drum, with fast release and medium or even slow attack, will give a bigger sound as the decaying signal is lifted to the level of the initial peak, creating a sustain. The acoustic will considerably affect the character of the sound, and is worth experimenting with.

Piano will come through well using a tight slope, medium/slow attack and fast release. The same goes for vocals in a rock group where high mean levels must be maintained to retain intelligibility. Some presence can be added after compression to help. More normally, the use of the softer slopes on vocals will retain expression and dynamic range. Compression with fast release will compensate for movement around the microphone.

Where direct injection is possible (for example, on bass, rhythm, lead and keyboard instruments), it can be worthwhile compressing the direct signal (to avoid spurious pickup) and mixing this with acoustic pickup.

Weaker instruments (like violins) can be given more body by compression; care must be taken, however, to watch out for pickup from foldback headphones. If this happens a good expander will maintain a clean track. With vocal or handclap overdubs, pickup from cans will be a problem; in this case impressive cleanup can be achieved with a gating action.

Gating or expanding the bass drum (depending on separation) can also be effective. A fast attack will give a sharp edge (like a stick), while slow attack will create a mellow, rounded 'leather pedal' sound. With fast release, the threshold should be adjusted until maximum cleanup is obtained.

When there is a lot of splash from cymbals (the mic was probably wrongly placed) it may be best to use a slower attack, so as to respond to the drum rather than the cymbal. A gate with a frequency conscious side-chain could be helpful.

Selective expanders or dynamic filters have proved to be useful. A highpass version can be used to attenuate low-level acoustic rumble or electronic hum, until sufficient wanted lf is present to mask it. At this point the system would be adjusted to give a flat response.

Similarly, a lowpass dynamic filter can attenuate electronic hiss or high-frequency splash around a bass instrument, or be used with guitar and keyboards—opening to give a flat response in the presence of wanted hf transients and signal. ▶

COMPRESSORS AND LIMITERS: THEIR USES AND ABUSES

Stage

Expanders can make a useful contribution to stage work on vocal mics and direct injection keyboards since often more are kept live than are being used at any one time.

In sound reinforcement situations, compression can considerably add to the effective power output of the system. This can either be achieved by a limiter alone, or a compressor-limiter combination. Overall limiting would be essential to protect amps and speakers. Care must be taken to allow for increasing gain on recovery, which will affect feedback levels.

Expanders with variable range control can be useful in recording speech against a high ambient noise level. In such cases it is probably best to accept some noise, rather than try to eliminate it all. This can be done by adjusting the range for approximately 10 dB attenuation with medium attack and fast release, and then setting the threshold to open on voice. Background noise will be masked by the voice and attenuated during pauses by 10 dB.

Classics

In classical recording where high-level compression causes a reduction in upper-level dynamic contrast, an alternative form of compression can be arranged.

When a limiter-compressor is placed in parallel with the direct signal, it is possible to obtain low level compression; the advantage being that the slope gets progressively softer as level rises, until finally returning to a 1:1 condition. In order to retain a correctly related internal dynamic balance between the original and compressed signal, it is essential to have a very soft slope with low threshold level. Compression commences just above the lowest signal level; this way the compressed signal can be a true reduction of the original.

Happily one of the effects of arranging the compressor in parallel is to soften even further the slope selected: for example, the 2:1 ratio is reduced to >1.5:1, while a 1.5:1 slope becomes 1.25:1 with a threshold of 60 dB down on peak level.

The procedure is as follows: adjust the direct signal for required peak output (if live signal it may be preferable to use a limiter on the final output); connect a compressor in parallel and select the lowest ratio available that will give 20 dB reduction; adjust the compressor to give 20 dB compression at peak input level; then set the peak output level of the compressor to be 10 dB below the peak level of the direct signal. The two signals are mixed and the effect will be approximately 12 dB overall compression.*

Although this is similar to the Dolby arrangement, it would be unwise to use Dolby units as single-ended compressors since there will be considerable spectral energy distortion due to the action of the band processors. Plus the drawback that since Dolby units use a limiter slope the ratio will be too tight.

Conclusion

A wide range of limiters and compressors are currently available to meet the many applications to which they are now put. Simpler devices, although in general easier to operate, must compromise on the range of options available, which in turn restrict their application in creative engineering.

By their very nature, units that offer greater flexibility require a higher degree of operational competence and discipline on the part of the engineer. It is essential that he understands what he is trying to achieve, and know what needs to be done to get the effect he is after. It is inevitable though, that in inexperienced hands combinations of such widely varying parameters could produce disappointing results.

It is a sad fact that, due to occupational pressures, many engineers just don't get time to fully explore the possibilities of their auxiliary equipment; for those who do, the rewards can be high. It is so often the ability to produce that little extra something that brings recognition by an artist and earns an engineer the accolade: *master of his art*.

**Detailed application notes are available from the author. A large sae would be appreciated.*