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COMPRESSION APPLICATIONS

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Introduction

The most powerful weapon in the audio arsenal is often the most misunderstood. Frequently, it is also the most misused. We are talking, of course, about compression - the signal processing technique that alters the dynamic range of an audio signal so that the loudest passages are reduced in level relative to the softest ones, making it easier to hear every nuance of the music. A compressor (or "limiter," when acting more severely) automatically "rides gain" on a signal much like a recording engineer does by hand as he manually raises and lowers the faders of a mixing console.

Properly applied compression tightens up mixes and makes for exciting, upfront vocals and the punchy bass and drums that are the foundation upon which great pop music is built. Compression can also add coloration, making a signal warmer and fatter or brighter and edgier. It can even be a musical tool, enhancing sustain or providing a snappier attack to selected instruments or individual notes.

On the flip side, you've no doubt already heard examples of poor applications of compression - whether you realize it or not. If a track is not compressed sufficiently, the result may be an indecipherable vocal with missing consonants or a bass line where only certain notes punch through and others are lost. A track that is overcompressed will stay pretty much at the same volume all the way through (making for a boring listening experience) or can result in obvious crunchiness in your TV speaker when a loud signal like a cymbal crash triggers a reduction in volume of the overall audio. Which isn't to say that too much compression is always a bad thing: One well-known producer/engineer typically records bass by compressing both the direct and the amplified signal individually. He then submixes the two signals to a bus and straps a third compressor across that bus, then returns the signal to the console before routing it out again in combination with the kick drum, applying a fourth compressor across the new combined signal. And this is an engineer that always places a stereo compressor across his mix bus - someone who also, he avows, even relies upon his mastering engineer to provide yet more compression. Sounds like overkill, right? But this is someone who has dozens of platinum records to his name.

In the world of compression and limiting - as in all audio recording - it's your ears that determine proper usage, not a set of rules. Creative compression techniques can open up a world of effects, from that special snap on a snare drum to an over-the-top room sound, to the characteristic distortion that can only come from overdriving a dynamics processor. That's where the real art of compression comes in, with most professional engineers developing their own pet applications, many of which we will share with you in these pages.

Why Compress?

The basic concept behind a compressor/limiter is relatively simple. It is a device in which gain is automatically adjusted in a predetermined ratio in response to changes in the input signal level: in other words, it keeps the volume up during softer sections and brings it down when the signal gets louder.

While the terms compression and limiting are often used interchangeably, it is generally accepted that when this kind of dynamic processing occurs at ratios below 10:1 (that is, where a change of 10 dB or less in input signal level results in an overall change in output level of 1 dB) it is referred to as compression; when higher ratios are used, it is known as limiting.

At lower ratios, compressors gradually reduce the dynamic content above a user-determined threshold level. At higher ratios, limiters abruptly prevent signals above the threshold level from exceeding a certain maximum value.

In multitrack recording, dynamics processing (another term for compression and limiting, as well as expansion and gating, which produce the opposite effect) is generally applied either to individual instruments or small groups of instruments. In mastering, and for radio and television broadcast, entire songs are compressed. Either way, the reasons for using a compressor range from the necessary to the creative.

Overload Protection

The original purpose of compressor/limiters was to protect equipment from excessive signals. Particularly in radio broadcast, the final power-amplifying tube in a transmitter - a large and expensive piece of equipment - was at risk. Even today, despite the advent and proliferation of digital recording (which provides the blessing of noise-free, wide dynamic range) it's all too easy to generate undesired distortion by overloading tape or circuits with uncontrolled signals. Ironically, the inherent limitations of digital technology make signal overloading even more of a problem, resulting in an especially harsh distortion known as clipping.

Overload protection is even more important in live sound applications when onstage performers are using in-ear monitors. In such an environment, a sudden, uncontrollably loud signal can cause permanent hearing damage. Live sound engineers rely on severe limiting (often called "brick-wall" limiting) to quickly and automatically reduce the level of signals exceeding a certain threshold.

Gain Consistency

As we have seen, a compressor acts like an automatic fader. It reduces the difference in volume, or gain, between the loudest and softest parts of a performance, a function that is necessary for many practical reasons.

The dynamic range of human hearing (that is, the difference between the very softest passages we can discern and the very loudest ones we can tolerate) is considered to be approximately 120 dB. Many of today's digital recording media approach that kind of dynamic range. However, unless the listener is in a virtually noise-free environment, quiet passages of recorded music can be lost in the ambient noise floor of the listening area, which, in an average home, is 35 to 45 dB.

Likewise, when program material with wide dynamic range is reproduced through a medium with narrow dynamic range, such as radio, much information is likely to be lost. Everyone has heard the ultimate combination of these problematic factors: listening to music broadcast over the radio while riding in a car with the windows open.

To prevent these problems, compression is customarily used to reduce the dynamic range of program material to a level appropriate for the environment in which it is to be heard, as well as for the medium through which it is to be reproduced.

During recording, compression is customarily used to minimize the volume fluctuations that occur when a singer or instrumentalist performs with too great a dynamic range for the accompanying music. It can also help to balance out volume fluctuations within an instrument itself, when, as commonly occurs, certain notes of a bass guitar resonate more loudly than others, or when a trumpet plays louder in some registers than in others. Properly applied compression will make a performance sound more consistent throughout.

Increased Loudness

When dynamic range is reduced, the overall level of the material can be raised (made "hotter") without the concern that peak levels will cause distortion. This kind of "squeezing" also serves to raise up the quiet sections of the program. By increasing the ratio of average-to-peak levels, compression enables a signal to be made significantly louder while the overall peak level of the material is increased only minimally.

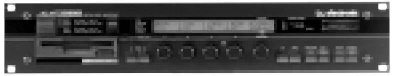
This application is particularly desirable in music that is destined to be played on the radio and television, where the competitive advantage of having your song stand out by being louder than others is obvious.

Increased loudness is also an important factor during recording, both to analog tape and to digital formats. When recording to analog tape, compression can help to raise the level of the signal being recorded to an optimum level above the noise floor of background tape hiss.

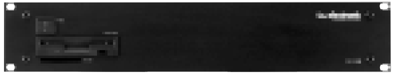
When recording to a digital medium, compression can help ensure that the signal is encoded at the highest possible level, where more bits are being used so that better signal definition is achieved.

Finalizer 96K

Triple•C



M5000



System 6000

Input

Sounds obvious, right? A knob that controls the amount of signal entering a device, an Input Level Control is found on most, but not all, signal processors. Its usage, however, is a bit more complicated in compression than in some other applications, because the amount of input directly affects the threshold at which a compressor/limiter begins to work. The actual structure of Input controls and where unity gain (i.e., no signal boost or attenuation) appears on the knob will vary among manufacturers; it's a good idea to look up what they recommend as a starting position. For example, the TC Electronic Finalizer's Analog Input offers gain up to +26 dB or attenuation to -6dB, while the Triple•C Input level knob ranges from -6 to +18 dB.

Threshold

Threshold is perhaps the most critical control of any compressor. It sets the level at which a compressor starts to work. Below the threshold point, the volume of a signal is unchanged; above it, the volume is reduced. For example, if Threshold is set to 0 dB, input signals at or above 0 dB will be compressed, while those below 0 dB will be unaffected.

The incoming signal may be detected one of two ways - either by viewing signal peaks or by RMS ("Root Mean Square," a formula for detecting average signal level, much like our ears do). A compressor may offer input detection by either means, or, like the TC Electronic Finalizer, Triple•C, M5000 and System 6000, it may allow a choice between the two. Generally, peak detection works best for applications where you are using a limiter to prevent signal overloads, while RMS detection works best when you are using a compressor to raise overall apparent level.

Ratio

The amount of increase in input signal (in decibels, or dB for short) required to cause a 1 dB increase in output is called the Compression Ratio. A ratio of 1:1 means that for every 1 dB of increase in input level, there is a 1 dB increase in output level; in other words, there is no compression being applied. But a ratio of 8:1 means that an 8 dB increase in input is required to produce only a 1 dB increase in output.

A compressor's Threshold and Ratio controls work in tandem. At a lower threshold setting, lower level signals are compressed. However, when the ratio is set lower - for example, 2:1 - the compressor has less effect on the signal. Ratios of 10:1 and greater are generally considered to be limiting as opposed to compressing. At high ratios of 20:1 or higher (some limiters even offer a theoretical infinite ratio of ∞:1), "brick wall" limiting kicks in - that is, any change in input, no matter how great, results in virtually no increase in output level.

Soft Knee vs. Hard Knee

A compressor's "Knee" is related to its Threshold control. The knee determines whether the compressor will reach the maximum selected amount of gain reduction quickly or slowly. A gradual transition ("Soft Knee") from no response to full gain reduction will provide a gentler, smoother sound, while a more rapid transition ("Hard Knee") will give an abrupt "slam" to the signal. Therefore, a soft knee is generally preferred for most program material, although the hard knee's attack can be useful in peak limiting and special situations such as de-essing.

The knee of most compressors is preset to either hard or soft, but some manufacturers offer the ability to choose between the two.

Attack

The key to the operation of any compressor is the setting of the Attack and Release times; these are the parameters which most affect how "tight" or how "open" the sound will be after compression.

Once a signal has crossed the threshold, the attack time is the amount of time it takes for the gain of a signal to decrease by the amount specified by the ratio control.

A fast attack kicks in immediately and catches transients, reducing their level and thus "softening" the sound. A slow attack time allows transient signals to pass the threshold unscathed before compression begins.

Beware: If an attack time is too short, desirable initial transients may be lost, softening the sound in an unintended way, perhaps even clipping consonants on a singer's voice.

Release

The Release time is the time it takes for the signal to return to its initial (pre-gain reduction) level after it drops below the threshold point.

If the release time is set too short for the program material, with too much gain being restored each time the signal falls below the threshold, "pumping" and "breathing" artifacts occur, due to the rapid rise of background noise as the gain is increased. If the release time is set too long, however, a loud section of the program may cause gain reduction that persists through a soft section, making the soft section inaudible.

Output (Makeup Gain)

Last in line is the Output Control, also known as "makeup gain" because it is used to make up for the gain reduction accomplished by the compressor. To make it easier to compare the result of the processing, the output control should be set to unity gain, so that the processed signal is raised to the point at which it matches the level of the unprocessed input signal (for example, if a signal is being reduced in level by approximately -6 dB, the output makeup gain should be set to +6 dB). Some compressors, such as the TC Electronic Finalizer, can automatically calculate makeup gain as a function of the threshold and ratio settings.

Bypass

Don't neglect the humble Bypass function - it's a tool that will let your ears decide if you are actually improving the signal! A switch which routes the input directly to the output, Bypass allows for direct A/B comparison of the processed and unprocessed signal.

Look-Ahead

Some compressors feature a "Look Ahead" feature, where, by slightly delaying the audio signal, the compressor can respond to the incoming signal before it is heard, thus enabling it to react more quickly and accurately to changes in level. The look-ahead delay time is usually scaled automatically with the attack time, but on some compressors, such as the compression algorithm in the TC Electronic System 6000, the delay time may be manually set by the user.

Sidechain Input

A Sidechain Input allows a compressor to respond to a signal other than the one that is actually being affected. Many compressors provide a sidechain insert that allows processing of the signal before it reaches the detector, a function useful for such frequency-sensitive functions as de-essing.

When de-essing, the high frequencies are boosted on the sidechain input signal, while the low and mid frequencies are cut. The compressor's detector reacts to the excessive highs, reducing the gain of that portion of the signal alone.

The opposite reaction - "ducking", or automatically lowering the level of a signal - may also be achieved through use of a sidechain input. This technique is often used in radio and television commercials to lower the level of a music bed while the announcer is speaking. The voiceover track is sent to the sidechain input, with the threshold set low enough to respond each time the voice is heard, causing the music to duck while the voice is speaking.

Stereo Linking

Generally used on stereo instruments, background vocal pairs or stereo mixes, a Stereo Link function allows two channels to be run through a compressor with both channels affected in the same manner by one common set of controls. This keeps the levels of the two sides even, but avoids problems with center shifting in the stereo image that may occur if separate compressors are used on each side. Stereo linking may work either of two ways: the channel receiving the most gain reduction may determine the gain reduction for the other channel, or it may be possible to designate one channel to be a master and the other a slave.

Full-Range Versus Multiband Compression

Traditional compressors operate over the entire frequency range of the incoming signal; that is, they respond to changes in input regardless of where in the frequency spectrum transients may be occurring. While there are certainly advantages to using such a circuit design (for example, when compressing a single source signal with a limited bandwidth, such as a snare or kick drum), more modern compressors - such as the TC System 6000, M5000, Triple•C, and Finalizer - offer a more flexible approach known as Multiband Compression, which is better suited to signal with broad frequency range. Here, the incoming audio signal is split into several user-determined frequency areas so that compression can be applied selectively to each. For example, if you are compressing an overall mix, a multiband compressor allows you to control low-frequency transients (such as kick drum hits) without affecting midrange and low frequencies, thus avoiding "pumping" or "breathing" signal artifacts. Specific examples of using both full-range and multiband compressors are given in Part III of this booklet.

Crossover (Multiband Compression)

As described in the previous section, multiband compressors such as the TC Electronic Finalizer, Triple•C, M 5000, and System 6000 make it possible to optimize dynamics separately over different frequency areas. Multiband compression avoids many of the problems inherent with traditional compression, such as the loss of high frequencies in compressed vocals.

The Crossover Control in multiband compressors allows the user to define the lower and upper limit of each of the frequency bands, allowing compression to be applied only to the necessary areas, and different compression parameters to be applied to different frequency bands. The crossover points must be selected carefully while closely listening to the program material, since even slight changes can alter the sound dramatically.

Envelope

The TC Triple•C compressor offers a special mode not found in any other compressor, called Envelope Mode. Here, the actual shape of the incoming signal can be altered by changing its level at the attack and release points. When the signal crosses the user-defined threshold, it is boosted or attenuated to the level defined by the Envelope Attack Gain parameter and then returns to the threshold level at a user-defined attack time. After the signal drops below the threshold level, a release parameter determines the amount of time it takes for the signal to decay before being boosted by the Envelope Release Gain parameter. Envelope mode is useful if you want to increase the apparent sustain of a sound or otherwise significantly alter its sonic characteristics (i.e., to create a more aggressive snare drum by boosting its attack level or to soften the hard transients of an acoustic piano or electric guitar).

In this section, we'll examine specific applications for dynamics processing. Compression and/or limiting can be applied to either single source signals (i.e. from individual live microphones or individual recorded tracks) or to multiple source signals (i.e. from submix buses or an overall stereo or surround mix). Let's look at each in turn.

I. SINGLE SOURCE SIGNALS

Vocals

Perhaps the most common application of compression to single source signals is to vocals, primarily because the human voice has an unusually wide dynamic range. Trained opera singers, for example, have been known to shatter glass at a few paces!

Full-range compressors tend to work well on vocals, since they do a good job of preserving legibility and articulation. However, multiband compression is sometimes preferable, especially if you want to control specific midrange transients that might otherwise trigger unwanted gain reduction of higher frequencies.

In live performance, Front of House (FOH) engineers almost always strap full-range compressors across vocal channels in order to maintain a consistent level even when the performer moves off-mic and to ensure that whispered and softly sung words are heard clearly while loud, shouted passages do not overload power amplifiers and PA speakers. In these cases, fairly low thresholds are set (often resulting in 10 dB or more of gain reduction) in conjunction with high ratios of 8:1 or greater. In order to maintain legibility, moderately slow attacks (5 - 25 ms) are used to allow transients (such as consonants at the beginnings of words) to pass through unscathed; release times are set to be as short as possible without pumping artifacts (typically, around 100 ms). Backing vocals are often submixed and compressors applied across the bus, with similar settings.

In the studio, vocal compression can be used much more subtly, especially if the singer has a trained voice and good mic technique. When recording vocals, ratios are generally 4:1 or less (sometimes as low as 1.5:1), with thresholds set so that there is only 3 - 6 dB of gain reduction on average. As with live performance, attack times must be sufficiently slow as to allow the beginnings of words to pass, with release times long enough to avoid obvious pumping (however, shorter attack and release times can be used to create a breathy "in-your-face" vocal effect). During mixdown, when there is more control, multiband compressors set to higher ratios of 6:1 or greater are often used to tame lead vocals, particularly when they need to sit in a dense bed of instrumentation. Mix engineers often apply different compression amounts to different passages (i.e., a softly sung verse may receive more compression than an aggressively sung chorus) or even to individual words. Backing vocal tracks are generally submixed in mono or stereo (see the "Stereo Signals" section below for more information on the latter), with more severe degrees of compression (i.e., lower thresholds, higher ratios, and shorter attack and release times) typically applied.

Another common vocal compression trick used during mixdown is to mult a vocal track so that it appears on two faders, severely compress one of the two, and then carefully mix the two signals together. This serves to preserve much of the original dynamic range while allowing softly sung sections to cut through more clearly.

De-Essing

Sibilance in a voice - those nasty bits of distortion caused by too much of the letter "s" (as well as fricatives like "f" and "t") - can be selectively reduced by the use of a de-esser. This is a frequency-dependent compressor that utilizes a filter in the input signal circuit (or in a sidechain circuit) so that compression is triggered when an excess of high frequency signal is detected. Multiband compressors are especially well-suited for this purpose; simply use the Crossover control to isolate the frequency band in which the offending "esses" are occurring (typically 7 - 10 kHz) and then compress that frequency area only.

Note that de-essers are normally patched before console and outboard equalizers so that some of the high frequencies removed by the de-essing process can be restored, albeit without the distortion and radical increases in level that were originally present.

Guitar

There are, of course, many different kinds of guitars, ranging from the overdistorted electric lead sound of heavy metal, to the clean, crisp sound of a direct injected (DI) funk guitar, to the gentle strumming of an acoustic guitar - but compression can aid in the sound of all of them.

For a heavy rock guitar, try a full-range compressor with a severe limiting ratio of 10:1 or even 20:1, with a threshold that yields 6 - 10 dB of gain reduction and a sufficiently long attack time so that the transients of each chord or note are unaffected. Clean funk guitars benefit from the use of multiband compressors, with a ratio of 6:1 on the midrange frequencies and a lesser ratio (3:1 or 4:1) on the high frequencies. Again sufficiently long attack times must be used so that the rising edge of each chord is clearly delineated. Steel-string acoustic guitars also benefit greatly from compression, either multiband or full-range, depending largely upon the quality of the guitar (a well-constructed instrument will yield a more consistent sound across all strings, and thus will work fine with a full-range compressor; a lower-quality guitar with apparent differences in level from string to string or even note to note will benefit from multiband compression). Here, gentler compression ratios of 1.5:1 or 2:1 are in order, with sparing use of the threshold control so that there is nominal gain reduction of 2 -3 dB; again, the attack time must be sufficiently long so as to not "soften" the sharp sound of the plectrum on the string.

Bass

Because it is important that every bass note be heard clearly (and because few basses are constructed well enough to deliver each note at equal level, nor are there many bass players who have sufficiently good technique to deliver this), basses are almost always compressed; many engineers even double-compress them - once during recording and again during mixing.

A typical starting setting for a full-range compressor affecting a rock bass would be to have a ratio of 3:1 or 4:1, with the threshold set to deliver 4 - 8 dB of gain reduction for peak notes. Hard knee compression works well here. If the bass is played with a plectrum, the attack time can be fairly short so as to smooth out the start of notes; if played with the fingers (and especially if played funk-style), the attack time should be somewhat longer so that transients get through unscathed. Release time is particularly important with bass, and should be sufficiently long so as to impart smooth sustain to held notes.

Multiband compressors work really well on bass, since they allow the lowest frequencies to be tightened up (reducing "woofiness") without affecting the mid- and high-midrange frequencies, which carry the attack of the sound. When compressing low frequencies alone, you can use very fast attack times since the wavelengths are so long; however, for the same reason, you'll need to set very long release times.

Drums

Drums are usually recorded with multiple microphones and often submixed to a stereo pair during mixdown. Typically, separate microphones are used to capture the sound of the snare drum, kick drum, each of the tom-toms, and sometimes the high-hat. Most often, a pair of overhead mics is used to capture the cymbals (and add some "air" to the drum kit), and, depending upon the studio environment, room mics may be set up as well to add ambience.

Compression is almost always applied to the snare and kick drum tracks so as to add "snap" to the signal and level out all the hits so that they are consistent throughout the song. Full-range compressors work fine for this purpose due to the limited frequency bandwidth of the snare and kick.

Typical settings for snare drum compression are a ratio of 3:1, with a gain reduction of 4 - 8 dB and very fast attack (so as to "squash" the sound) and release times. The kick drum will have similar settings, but with slightly higher ratio (4:1 or 5:1) and a slightly lower threshold (set for gain reduction of 6 - 10 dB). Soft knee compression works best for both applications, and careful setting of the attack time is critical to achieve the effect you want. Tom-tom tracks are also sometimes compressed, with similar settings.

Compression is also typically applied to drum kit room ambience mics as an effect. Here, much higher limiting ratios (10:1 or even 20:1) are applied, with low thresholds (8 - 12 dB of gain reduction) and very slow attack and very fast release times so as to engender obvious pumping. When mixed in carefully with the close-miked tracks (and especially if the ambient tracks are sharply gated), this can serve to impart the huge "room" sound prevalent in many rock records.

Piano

The acoustic piano (short for "pianoforte," which literally means "soft-loud") is capable of producing a staggeringly large dynamic range. This is one reason why it works so well as a solo instrument, and in classical or jazz recordings, piano is rarely compressed at all because the beauty in the music lies in the very dynamic range of the performance.

However, in rock music, or wherever piano appears as an ensemble instrument, gentle compression can help balance out the sound, particularly when certain notes or chords jump out. Here, multiband compressors work extremely well because you can home in on the precise frequency area that needs to be tamed without affecting the rest of the range of notes. Gentle ratios of 1.5:1 or 2:1 are often used (perhaps as much as 4:1 in rock music), with threshold set so that there is no more than 2 - 4 dB of gain reduction. The attack time is particularly critical since it is vital that the transients - the attack of the hammer on the strings - be preserved; usually it is set no faster than 50 msec.

The release time also has to be carefully set so as to avoid pumping; however, where sustained notes are played, longer release times should be used.

Organ

Rock organ produces sustained notes with slow attacks, and compression can often help smooth out a performance, especially when the organist is not sufficiently schooled in the use of the volume pedal. Here, multiband compressors work best since they allow you to identify the particular frequency range that needs dynamic processing. High limiting ratios of 10:1 or 12:1 are often used, with low thresholds that result in 8 - 12 dB of gain reduction. Since there really is no transient in most organ sounds (other than when the percussion switch is engaged), short attack times of a millisecond or less work quite well; release times must be sufficiently long so as to provide sustain without audible pumping.

If a multiband compressor is not available to you, you can try this trick: simply apply a full-range compressor to the microphone on the low Leslie speaker, and leave the other mic (which is picking up the rotating horn carrying the high frequencies) uncompressed. This will serve to effectively compress the low frequencies only, giving you the ability to tighten up the sound without altering the attack transients.

II. MULTIPLE SOURCE SIGNALS

Stereo and Dual Mono Signals

When compressing stereo signals (i.e., acoustic pianos recorded with two microphones, backing vocals panned across a stereo soundstage, or ambient tracks recorded with stereo mics), it is generally desirable that the stereo link switch on the compressor be engaged so that transients in one channel do not trigger gain reduction in the other channel, since this will result in audible "pumping" and "breathing" artifacts.

When compressing dual mono signals, however, the stereo link switch can provide some interesting creative effects. One trick used by engineers back in the '60s was to route a mono backing track into one channel of a stereo compressor and then mult the lead vocal, using one mult to mix its level in with the backing track, and feeding the other mult into the input of other channel of the stereo compressor but without taking an output from that channel. The threshold of the lead vocal channel would be carefully set and the stereo link button engaged, so that whenever the vocalist was singing, the backing track would be slightly reduced in volume, allowing the vocal to better "sit."

Another trick in a similar vein is to take both a bass DI (Direct Inject) signal and a miked bass amplifier and record each signal on a separate track. Then feed each into a channel of a stereo compressor and engage the stereo link switch. This way, on the notes where the DI signal, for example, might be louder than the bass amp signal, the compressor will control the level of both simultaneously.

Interesting effects can also be created by compressing stereo reverb returns and/or sends. By using long attack and release times (with fairly severe ratios of 8:1 or greater), this will serve to lift the level of the reverb whenever the input signal starts to die away, making for an interesting (if unnatural) "in-your-face" sound.

Overall Bus Compression

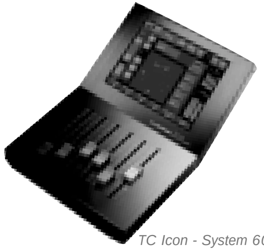
"Compression is the sound of rock," observes one multiplatinum engineer. Certainly it is the sound of radio and television; virtually all broadcast facilities strap compressors or limiters across the final mix bus. Back in the days of vinyl, mastering engineers also typically applied compression to the overall signal so as to maximize level and tame transients that might cause the stylus to literally jump out of the grooves.

Today, in the era of the CD, such applications of compression are no longer strictly required in a technical sense; however, record company marketing executives inevitably strive to produce the loudest recordings possible on the theory that louder not only sounds better but also better captures the attention of the listener. As a result, not only do mastering engineers continue to apply liberal doses of compression to the finished two-track tapes or disks arriving at their facility, but mixing engineers often apply it beforehand, strapping a compressor across the stereo mix bus.

In these kinds of applications, however, less is more, particularly since overcompression is difficult if not impossible to remove. Typical threshold settings are very low so that only 1 - 2 dB of gain reduction is occurring, and only when severe peaks occur. Ratios are also very gentle, in the 1.5:1 to 2:1 range, with long attack and release times (50 msec or greater for the attack; 250 msec. or greater for the release) so as to avoid pumping.

The advent of DVDs and multichannel audio means that there will be times when a multichannel compressor is required to affect all channels in a coherent way. The TC System 6000 offers such an algorithm - the MD-5.1 - which is widely used by mixing and mastering engineers working in the rarified field of surround sound. This provides five discrete channels of 3-band compression as well as a full-range brickwall limiter on all outputs, plus a separate LFE (Low Frequency Effects, or subwoofer) channel with full-range compression and limiting. Most significantly, three assignable sidechain link inputs for each of the five main channels allow selective and interactive compression of multiple sources without pumping artifacts. For example, the two rear and two front channels can be linked to one another, with the center channel responding to changes in level occurring from both center and front left and right feeds.

Because surround sound enables the overall mix to be spread among six, as opposed to two, speakers, less compression (and equalization) is generally required than in traditional stereo mixes. Compression ratios for surround signals are therefore generally very low - 2:1 or 3:1 - with long attack and release times and thresholds set so that there is rarely more than 3 - 5 dB of gain reduction.



TC Icon - System 6000

Compressing The Headphone Cue Mix

When recording vocals, the objective is to get the singer to deliver as strong a performance as possible while still singing on pitch. To this end, the headphone cue mix is extremely important, and many singers prefer to hear the song in their headphones in as finished a state as possible. For this reason, many engineers opt to strap a compressor or limiter across the auxiliary send(s) feeding the headphone amplifier, often setting the threshold sufficiently low and release time sufficiently fast so as to impart an audible pumping, same as the track might sound when played on the radio. Some even use multiple dynamics processors for this purpose, strapping a mono compressor across the vocal track and a stereo limiter across the backing track which is triggered by a sidechain input from the vocal so as to impart a subtle "ducking" whenever the vocal enters. The bottom line, as one producer says: "Whatever the vocalist likes to hear that gets them to deliver a performance, that's what I give them."

Brickwall Limiting With In-Ear Monitors

As noted in Part I of this booklet, the usage of in-ear monitors during live performance provides performers with an accurate, highly controllable means of hearing themselves without having to rely on onstage wedge speakers which are prone to feedback. However, an uncontrollably loud signal routed to an in-ear monitor can cause permanent hearing damage, so live sound engineers rely on severe limiting (often called "brick-wall" limiting) to quickly and automatically reduce the level of signals exceeding a certain threshold. In such an application, the ratio is set as high as possible: 10:1, 20:1, or even, if available, ∞:1 (infinity to one), with the threshold carefully set to a point well below that at which hearing damage can occur. The attack time is generally very fast so that high-level transients cannot pass through unaffected, with the release time set according to the program material.