CHAPTER 10

How to Manipulate Dynamic Range for Fun and Profit

PART TWO: DOWNWARD PROCESSORS

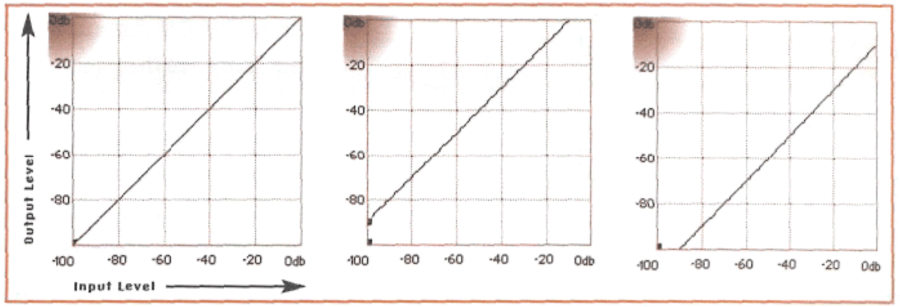
I. Compressors and Limiters: Objective Characteristics

Part two and Part three of this series are about microdynamic manipulation, which is primarily achieved through the use of dedicated **dynamics processors**. In this chapter (part two), we look at how *downward* processors work. Before we can learn how to use devices such as compressors and expanders, we must study the objective characteristics of the devices which perform the job.

Transfer Curves (Compressors and Limiters)

Let's begin with the **measurable characteristics** of processors which perform downward compression, simply called **compressors and limiters**.

A transfer curve is a picture of the input-to-output gain characteristic of an amplifier or processor. A straight wire or unity-gain* amplifier would yield a straight diagonal line across the middle at 45°, called the *unity gain line*. A family of **linear curves** can be drawn, as in these three figures:

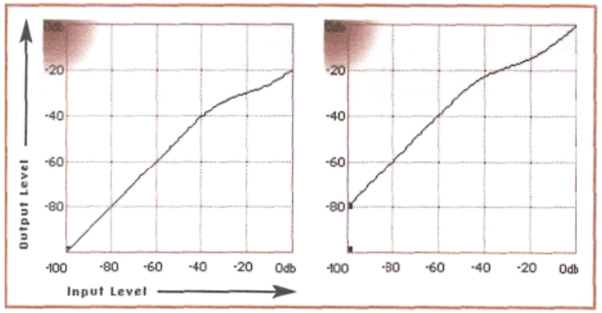


Three transfer curves. *At left, a Unity-Gain Amplifier, then an amplifier with 10 dB gain, then with 10 dB loss (attenuation).*

* Unity-gain means the ratio of output to input level is 1, or 0 dB.

Input level is plotted on the X axis, and output on the Y. At left is a unity gain amplifier, followed by one with 10 dB gain, and with 10 dB loss (attenuation). As long as there is a straight line (not a curve) at 45°, the amplifiers are linear. Notice that the middle plot would yield distortion for any input signals above –10 dBFS.

The threshold of a compressor is defined as the level above which gain reduction begins to occur. **Compression ratio is the ratio of input change to output change above the threshold**. At left in the following figure is a simple compressor with a fairly gentle 2.5:1 compression ratio, and a threshold at around –40 dBFS (which is quite low and would yield strong compression for loud signals). 2.5:1 means that for a level increase of the source of 2.5 dB, the output will only go up 1 dB, or for a rise of 5 dB, the output will only go up 2 dB, or as can be seen in the plot, an input change of 20 dB yields an output change of a little less than 10 dB (once the curve has reached its maximum slope). A compressor such as this would actually make loud passages softer, because the output is less than the input above threshold; this is always the case unless you follow the compressor with a gain makeup amplifier.

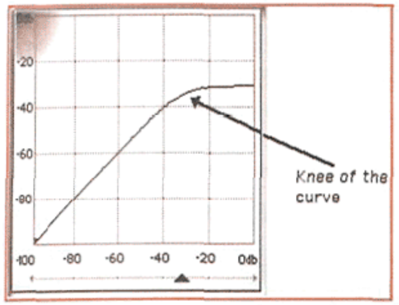


At left, Compressor with 2.5:1 ratio and –40 dBFS Threshold and no gain makeup. At right, the same compressor with 20 dB gain makeup.

At the right-hand side of the figure, by using gain makeup (a simple gain amplifier after the compression section), we can restore the gain such that a full level (0 dBFS) signal input will yield a full level signal output. In this illustration, the amplifier has an extreme amount of gain, 20 dB, which would considerably amplify soft passages (below the threshold). In typical use, makeup gains are rarely more than 3 or 4 dB. Loud input passages from about –40 to about –15 are still amplified in this figure, but above about –15 dBFS, the curve slopes back to unity gain and resembles that of a linear amplifier. Far below the threshold, it's a fairly linear 20 dB amplifier and can have pretty low distortion because there is no gain reduction action. At full scale, 20 dB of gain makeup is summed with 20 dB of gain reduction, yielding 0 dB total gain. This particular compressor model's curve levels off towards a straight line above a certain amount of compression, so the ratio only holds true for the first 15-20 dB above the threshold. Other compressor models continue their steep slope, thus maintaining their ratio far above the threshold. There are as many varieties of compression shapes as there are brands of compressors, and they all give different sounds. To get the greatest esthetic effect from any compressor, most of the music action must occur around the threshold point, where the curve's shape is changing; thus, it is likely a real-world compressor's threshold would be nearer –20 to –10 dBFS, where most of the musical movement takes place.

The following figure shows a very high ratio of 10:1, without gain makeup. Notice that the output is almost a horizontal line above the threshold. Most authorities call any compressor with a ratio of 10:1

or greater a limiter. There are very few analog compressors with greater ratios, however, some digital limiters have been built with ratios of 1000:1 in order to prevent even the minutest excursion or overload above full scale (0 dBFS). The portion of the curve at or near the threshold is called the **knee**, which is the transition between unity gain and compression. The shape of the knee can make the transition gentle, or hard. The term **soft knee** refers to a rounded knee shape, and **hard knee** to a sharp shape, where the compression or limiting kicks in quickly above the threshold. Conceivably, the change from unity to 10:1 could be instantaneous, in which case the knee would be a sharp angle instead of round, producing a sharp sonic change, thus a limiting effect. The need for a gentle knee depends a lot on how much musical activity is occurring at the threshold. If there is a lot of musical activity or movement around the threshold, the knee shape can be critical. For those models of compressors that do not have knee adjustments, some of the effect of the knee can be accomplished by tweaking the ratio and/or threshold.

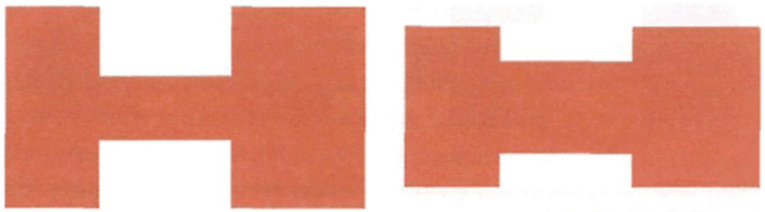


Compressor with 10:1 ratio, -32 dBFS Threshold, without gain makeup

Attack and Release Times

Attack time is defined as the time between the onset of a signal that is above threshold and full gain reduction. It can be measured in micro or milliseconds though it can be as long as a second or two. Typical compressor attacks used in music range from 50 ms to 300 ms, with the average used probably 100 ms. **Release time**, also known as **recovery time**, is defined as the time between when a signal drops below threshold and when the gain returns to unity. Typical compressor release times used in music range from 50 ms to 500 ms or as much as a second or two, with the average used probably 150-250 ms.* The terms **short** or **fast** with attack or release time may be used interchangeably, they mean the same thing. Similarly, **slow** and **long** attack and release times mean the same.

At the left side of the following figure is the envelope shape of a simple tone burst, from a high level to a low one and back again.

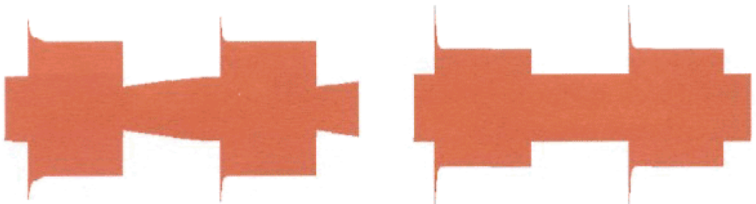


At left, a simple tone burst from high to low level and back. At right, the same tone burst passed through a compressor with very fast attack, high ratio, and fast release time

At the right side is the same tone burst passed through a compressor with a very fast attack, high ratio, and fast release, and whose threshold is midway between the loud and soft signals. Note that the loud passages are instantly brought down, the soft passages are instantly brought up and there is less total dynamic range, judging by the relative vertical heights (amplitudes).

* One manufacturer, DBX, measures release time in dB/second, which is probably more accurate, but I find hard to get used to.

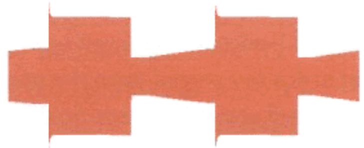
At left in this next figure is the envelope of a compressor with a low ratio, slow attack time and a slow release time. Notice how the slow attack time of the compressor permits some of the original transient attack of the source to remain until the compressor kicks in, at which point, the gain reduction brings the level down. Then, when the signal drops below threshold, it takes a moment for the release time to take action, and the gain is still low, then slowly the gain comes back up. A lot of the compression effect (the "sound" of the compressor) occurs during the critical release period, since as you can see, except for the attack phase, the compressor has actually reduced gain of the high level signal.



At left, a Compressor with a low ratio, slow attack time and slow release time. At right, higher ratio, faster attack and very fast release.

Contrast this with the compressor at the right, which has a much higher ratio, faster attack, and very fast release time. The higher ratio clamps the high signal down farther, and with the fast release, as soon as the signal goes below threshold, the release time aggressively brings the level up. This type of fast action can make music sound strongly compressed because it brings down the loud passages and quickly brings up the soft passages.

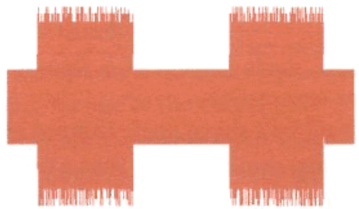
Here is another variation, a compressor with a release delay:



Output of a Compressor with a low ratio, slow attack time, slow release time plus release delay

A release delay control allows more flexibility in painting the sound character. Very few compressors provide this facility. It's useful when we want to retain more of the natural sound of the instrument(s), not exaggerate its sustain when the signal instantly goes soft, or reduce "breathing" or hissing effects when the source is noisy. The release delay is part of the subtle pastel color palette of the mastering artist.

The next figure illustrates what happens when the attack and release times are much too fast.



When the combination of attack and release times are extremely fast (typically <50 ms), a compressor can produce severe distortion, as it tries to follow the individual frequencies (waves) instead of the general envelope shape of the music

The distortion is caused by the compressor's action being so fast that it follows the shape of the low frequency waveform rather than the overall envelope of the music. This problem can occur with release times shorter than about 50 ms and correspondingly short attack times.

II. Microdynamic Manipulation: Adjusting the Impact of Music with a (downward) Compressor

The Mixing Engineer as Artist

Compressors, expanders and limiters form the foundation of modern-day recording, mixing and mastering. With the right device you can make a recording sound more percussive or less percussive, punchy or wimpy, smooth or bouncy, good or bad, mediocre or excellent.

When used by skilled hands, compression has produced some of the most beautiful recordings in the world, and a lot of contemporary music genres are based on the sound of compression, both in mixing and mastering, from Disco to Rap to Heavy Metal. A skilled engineer may intentionally use creative compression to paint a mix and form new special effects; this *intended distortion* has been used in every style of modern music. The key words here are *intent and skill*. Surprisingly, however, some engineer/artists don't know what uncompressed, natural-sounding audio sounds like. While more and more music is created in the control room, I think it's good to learn how to capture natural sound before moving into the abstract. Picasso was a creative genius, but he approached his art systematically, first mastering the natural plastic arts before moving into his cubist period. Similarly, it's good practice to know the real sound of instruments. Try recording a well-balanced group in a good acoustic space with just two mikes; it's a lot of work, and a lot of fun! Before multitracking was invented, there was much less need for compression, because close miking exaggerates the natural dynamics of instruments and vocals. At first, compressors were used to control those instruments whose dynamics were severely altered by close miking, e.g. vocals and acoustic bass. Later, when modern music began to emphasize rhythm, many instruments began to get lost under the energy, inspiring the creative possibilities of compressors and a totally new style of recording and mixing. Certainly the advent of the SSL console, with a compressor on every channel, changed the sound of recorded music forever.

Limiting Versus Compression In Mastering

Mastering requires new skills to be developed since we generally work on overall mixes instead of individual instruments. In mastering as well as mixing, compression and limiting change the peak to average ratio of music, and both tools reduce dynamic range. Most mastering engineers use compressors to intentionally change sound and limiters to change sound as little as possible, but simply enable it to be louder.* That's why limiters are used more often in mastering than in mixing. There is no perfectly invisible limiter, but compression changes the sound much more than limiting does. Think of compression as a tool to change the inner dynamics of music. While reducing dynamic range, it can "beef up" or add "punch" to low- and mid-level passages to make a stronger musical message. With limiting, however, with fast enough attack time (1 or 2 samples), and a

* As with compressors, it is the gain makeup process that permits the output of a limiter to be louder. When the peaks have been brought down, there is room to bring the average level up without overloading.

carefully-controlled fast release,* even several dB of limiting can be transparent to the ear. *Consider limiting* when you want to raise the apparent loudness of material without severely affecting its sound; *consider compression or upward expansion* (see next Chapter) when the material seems to lack punch or strength or rhythmic movement.

The BBC performed research in the 1940's demonstrating that distortion shorter than about 6-10 ms is fairly inaudible, which was the basis for the 6 ms integration time of the BBC PPM meter. In this modern solid-state world, some transient distortion as short as 1 ms will change the audible sound of the initial transient, particularly for instruments such as piano. So be sure to use your ears before limiting or reducing even short transients. With good equipment and mastering technique, wide range program material with a true peak to average ratio of 18 to 20 dB can often be reduced to about 14 dB with little effect on the clarity of the sound. That's one of the reasons 30 IPS analog tape is desirable as the medium to mix to: it has this limiting function built-in. A rule of thumb is that short duration (a few milliseconds) transients of **unprocessed digital sources** can be reduced by 4 to 6 dB with little effect on the sound; **however, this cannot be done with analog tape sources,** which have already lost the short duration transients. Any further transient reduction by compression or limiting will not be transparent (though it may still be esthetically acceptable or even desirable).

All digital limiters affect the sound to some extent, softening the transients and even fattening the sound slightly, as they allow us to raise the average level and the loudness. The less limiting we use, the cleaner and more *snappy* the sound, unless we are looking for a sound with softer transients. In an ideal mastering session, the limiter should only be acting on occasional inaudible peaks. Limiting distortion is especially audible on material which already has little peak information because a limiter is not designed to work on the RMS portion of the music and limiters can sound pretty ratty when pushed into the RMS region. Watch out for severe bass distortion because the time constants of a limiter are too fast for optimal compression.

A manual for a certain digital limiter reads "For best results, start out with a threshold of –6 dBFS." This is like saying "always put a teaspoon of salt and pepper on your food before tasting it." Instead, mastering engineers should judge how much limiting to use based on the desired absolute loudness (compared with other CDs) and how much degradation we can accept. Some sources can tolerate 6 dB of limiting without significant degradation, others 1 or none.

The World's Most Transparent Digital Limiter

***The most transparent limiter is to use no* limiter at all! When we are trying to make a section louder**, if there is a very short peak (transient) overload, for example, during a section

* The faster the release time, the greater the distortion, which is why the only successful limiters which use extra fast release times have **auto release control**, which slows down the release time if the duration of the limiting is greater than a few milliseconds. The effective release time of an auto-release circuit can be as short as a couple of milliseconds, and as long as 50 to 150 milliseconds. If limiting a very short (invisible) transient, the release time can be made very short.

of a drumbeat, a skilled mastering engineer can perform a short-duration gain drop that can be invisible to the ear, with the DAW's editor. This **manual limiting** technique allows us to raise a song's apparent loudness without the attendant distortion of a digital limiter, so it is the first process to consider when working with open-sounding music that can be ruined by too much processing. We can often get away with 1 to 3 dB manual limiting typically for a duration of less than 3 ms. But longer duration gain drops will affect the sound as much as or more than a good digital limiter. We use as little gain reduction as possible and when trying to make material louder, squeeze as much level as possible without clipping, for it helps keep the limiting invisible.

Equal-Loudness Comparisons

Since loudness has such an effect on judgment, it is very important to make comparisons at equal apparent loudness. During an instant A/B comparison the processed version may seem to sound better, if it is louder, but long-term listeners prefer a less fatiguing sound which "breathes." When you make comparisons at matched apparent loudness, you may be surprised to discover that the processing is making the sound worse, and it was all an illusion.

The Nitty-Gritty: Compression in Music Mastering

Consider this rhythmic passage, representing a piece of modern pop music:

shooby dooby doo **WOP**...
shooby dooby doo **WOP**...
shooby dooby doo **WOP**

The accent point in this rhythm comes on the backbeat (**WOP**), often a snare drum hit. If we strongly compress this music piece, it might change to:

SHOOBY DOOBY DOO WOP...
SHOOBY DOOBY DOO WOP...
SHOOBY DOOBY DOO WOP

This completely removes the accent feel from the music, which is probably counterproductive.

A light amount of compression might accomplish this...

shooby dooby doo **WOP**...
shooby dooby doo **WOP**...
shooby dooby doo **WOP**

...which could be just what the doctor ordered for this music because strengthening the sub accents may give the music even more interest. Unless we're *trying* for a special effect, and purposely creating an abstract composition it's wrong to go against the natural dynamics of music. (Like the TV weatherperson who puts an accent on the wrong syllable because they've been taught to "punch" every sentence: "The weather **FOR** tomorrow will be cloudy"). Much of hip hop music, for example, is intentionally abstract—anything goes, including any resemblance to the natural attacks and decays of musical instruments.

To manipulate the music requires careful adjustment of threshold, compressor attack and release times. If the attack time is too short, the snare drum's initial transient could be softened, losing the main accent and defeating the whole purpose of the compression. If the release time is too long, then the compressor won't recover fast enough from the gain reduction of the main accent to bring up the subaccent (listen and watch the bounce of the gain reduction meter). If the release time is too fast, the sound will begin to distort. If the combination of attack and release time is not ideal for the rhythm of the music, the sound will be "squashed," and louder than the source, but "wimpy loud" instead of "punchy loud." It's a delicate process, requiring time, experience, skill, and an excellent monitor system.

The best place to start adjusting a compressor is to find the approximate threshold first, with a fairly high ratio and fast release time. Adjust the threshold until the gain reduction meter bounces as the "syllables" you want to affect pass by. This ensures that the threshold is optimally placed around the musical accents you want to manipulate, the "action point" of the music. Then reduce the ratio to very low and put the release time to about 250 ms to start. From then on, it's a matter of fine tuning attack, release and ratio, with possibly a readjustment of the threshold. The object is to put the threshold in between the lower and higher dynamics, so there is a constant alternation between high and low (or no) compression with the music. Too low a threshold will defeat the purpose, which is to differentiate the "syllables" of the music; with too low a threshold **everything will be brought up to a constant level**.

Typical Ratios and Thresholds

When working on microdynamics in the above fashion, compression ratios most commonly used in music mastering are from about 1.5:1 through about 3:1, and typical thresholds in the –20 to –10 dBFS range. But there is no rule; some engineers get great results with ratios of 5:1, whereas a delicate *painting* might require a ratio as small as 1.01:1 or a threshold of –3 dBFS. Sometimes a recording requires the most gentle *invisible* compression without trying to alter its built-in dynamics. One trick to compress as invisibly as possible is to use an extremely light ratio, say 1.01 to 1.1 and a very low threshold, perhaps as low as –30 or –40 dBFS, starting well below where the action is. We may choose a low ratio to lightly control a recording that's too *jumpy* or to give a recording some needed *body*. It's unusual to see such low ratios used in tracking and mixing but very common in mastering of full program material, partly because with full program material, larger ratios may draw attention to the magic behind the curtain or reveal breathing, pumping or other artifacts.

We have noted before that every brand of processor (both compressors and expanders) has its own unique characteristics and sound. Part of the fun of mastering (and mixing) is discovering the special characteristics of different compressors. Even with the same settings, some are *smooth*, others are *punchy*, some bring out percussion better than others. This is not due to attack and release times per se, but rather to the curve or acceleration of the time constants, whether the device recovers linearly from gain reduction, whether the gain returns to unity quickly or slowly at the beginning. Design engineers spend much research time psyching out these particular characteristics, and the best we poor mortals can do is listen and see what we like.

Fancy Compressor Controls

Some compressors provide a **crest factor** control, usually expressed in decibels, or a range from RMS (or full average) to quasi-peak through to full peak. What this means is that the compressor acts on either the average parts of the music, the peak parts, or somewhere in between. Ostensibly, compressors with RMS characteristics sound more natural as they correspond with the ear's sense of loudness, but the best-sounding compressor I own is peak-sensing.

The Weiss model DS1-Mk2 is the first dynamics processor I've encountered with **two different release time** constants, *release fast* and *release slow*. The user sets a threshold of average transient duration, such as 80 ms, above which a sound movement is called *slow*, and below which it is called *fast*. Thus, instantaneous transients can be given a faster release time, but sustained sounds a slower one, which results in a more natural-sounding compression, especially with heavy compression. Indicator lights on the front panel aid in these adjustments.

Compression and Monitoring

I recall mixing a purist jazz recording using excellent powered monitors equipped with a driver protection circuit, which is ostensibly inactive except on peaks. However, when I arrived at my mastering room, I discovered that the recording "jumped out" too much, and required a bit of compression, a fact hidden during the mix and which I feel would have been similarly hidden had I monitored the mix with low-powered tube amplifiers (which self-compress).

As I mentioned in Chapter 6, it is a myth that you have to "precompress" for small systems. It's actually the converse. I made an excellent *snappy-sounding* master where we were concerned that the upper dynamics might have a bit too much upward impact. But when the recording was auditioned on a typical boom box or bookshelf system, the peaks were squashed compared to the mastering room audition and actually would have benefited from even more impact. Thus I have learned that if it "sticks out a little too much" on a high-headroom mastering system, then it's probably going to be fine when played on an inferior system. However, you'll never learn if something needs a bit more compression or is too compressed when listening on a monitor system that squashes the sound.

MYTH:
Program Compression is required to protect small reproduction systems.

Multiband processing

Multiband compression is probably the most powerful and potentially deadly audio process that's ever been invented. Basically, a multiband processor splits the information into two, three or more frequency bands, so that the compression action in one band will not cause another band to be affected. For example, if the vocal causes a bit of gain reduction, it will not pull down the bass drum (or vice versa), which might occur if you used a full-band compressor. This is the virtue and the justification of splitting processing into multiple bands. However, multiband compression has been overused, and hyped in my opinion. It can easily produce very unmusical sound or take a mix where it doesn't want to be. This tool requires careful judgment on the part of the mastering engineer.

Multiband processing was probably first introduced by TC Electronic in their M5000, then in their ubiquitous Finalizer, and brought to great sophistication (and much better sound quality) in their System 6000. Tube-tech has produced a three-band tube compressor. But multiple bands are hardly needed; one or two bands are usually enough. Rarely do even hip-hop recordings need more than two bands to sound punchy and strong. I use more than two bands in my mastering no more than a few times a year, when multiple bands have been a lifesaver. I largely use multiband compression (and expansion) to fix bad mixes that could not be remixed, for one key to a great master is to start with a great mix!

> *"One key to a great master is to start with a great mix."*

When To Consider multiband processing

- When there is a heavy and somewhat isolated bass drum and/or bass, splitting the processing into two bands prevents the drumbeats from modulating the rest, or vice versa.
- When you want to let transients (percussive sounds) through while still *punching* the sustain of the sub accents or the continuous sounds. Transients contain more high frequency energy than continuous sounds, so splitting the processing into a low and a high band permits using gentler compression or no compression at high frequencies (e.g., higher threshold, lower ratio).
- When there is too much sibilance. Sibilance can be controlled by using selective compression in the 3 through 9 kHz range (the actual frequency has to be tuned by listening to the vocalist). Try a very fast attack and medium release and a narrow bandwidth for the active band.
- When the mix is bad or certain elements appear to be weak in the mix, multiband processing can save the day, assuming a remix is not possible. I once received a rap project that was somehow mixed with very low vocal and extremely loud percussion and bass drum, and a remix was not possible. By compressing and then raising the level of the frequencies in the vocal range (circa 250 Hz) I was able to *remix the piece* and very nicely, turn the vocal up. Clearly, multiband compression is a power that should be used very wisely!

However, before trying multiband, first

- See if simply raising the attack time in a one-band compressor permits sufficient transient energy to come through. Or, try upward expansion (described in the next Chapter) instead.
- Try using few bands, only two if possible. This avoids potential phase shift and unnatural relationships between the mix elements of the mix, which can become the enemy of the mix engineer's delicate creation.

Equalization or Multiband Compression?

When multiband processing is available, the line between equalization and dynamics processing becomes nebulous, because the output levels of each band form a basic equalizer. Use plain equalization when instruments at all levels need alteration. Or consider multiband compression, to provide spectral balancing at different levels. For example, a song may get harsh-sounding when it gets loud, and it is possible to simulate the euphonic high-frequency saturation characteristics of analog tape by using a bit more compression at high frequencies.

If we're already using split dynamics, we make our first pass at equalization with the outputs (makeup gains) of each band. Multiband compression and equalization work hand-in-hand. Tonal balance will be affected by the crossover frequencies, the amount of compression, and the makeup gain of each band. In general, the more compression, the duller the sound, because of the loss of transients. I first try to solve this problem by using less compression, or altering the attack time of the high-frequency compressor, and as a last resort, I use the high frequency band's makeup gain or an equalizer to restore the high-frequency balance.

Clipping, Soft Clipping and Oversampled Clipping

Clipping is the result of attempting to raise the level higher than 0 dBFS, producing a square wave, a severe form of distortion. Clippers are devices which electronically cut momentary peaks out of the waveform to allow the overall level to be raised. Soft clipping attempts to do this with less distortion. I've decided that I don't like the quality of distortion produced by clipping or soft clipping, at least at 44.1 kHz SR (see Chapter 16). I believe there are better approaches. The first is not to raise the level at all, for many CDs are already too hot for their own good. Or use a good limiter, which sounds better than clipping to my ears. In Appendix 1, radio gurus Bob Orban and Frank Foti explain why clipping is a severe problem for radio processors. The jury is still out when it comes to oversampled clipping, whose distortion artifacts can be reduced by half in the audible (20-20 kHz) range, but isn't that really like saying *she's a little bit pregnant?*

Compression, Stereo Image, and Depth

"Not every instrument should be up front."

One sure way to destroy the depth in a recording is to compress it too much. Compression brings up the inner voices in musical material. Instruments that were in the back of the ensemble are brought forward, and the ambience, depth, width, and space are degraded. But not every instrument should be "up front". Pay attention to these effects when you compare processed vs. unprocessed and listen for a long enough time to absorb the subtle differences. Variety is the spice of life. As always, make sure the cure isn't worse than the disease.

The Mastering Engineer's Dilemma

Without compressors in CD changers and in cars, it is extremely difficult for the mastering engineer to fulfill the needs of both casual and critical listeners. It is our duty to satisfy the producer and the needs of the listeners, so we should continue to use the amount of compression necessary to make a recording sound good at home. But try to avoid using more compression than is

> *"Never in the history of mankind have humans listened to such compressed music as we listen to now."* — Bob Ludwig*

required for home listening. This approach will actually help radio play (see Appendix 1). If compromises have to be made for car or casual play, try to use transparent-sounding techniques such as parallel compression (see next Chapter), which satisfy even critical listeners. Audition test masters in all environments, hopefully arriving at a decent compromise.

III. For the Mixing Engineer: How To Avoid Hypercompression† during Mixing and Tracking

Letter from a DIGIDO.COM visitor:

> **I found your site through a link. I was looking for information on how to use my compressors to make my music better. What I found was instruction on how not to use my compressors to *make my music better. The quality of* my recordings has gone up greatly since I read your articles.**

How to Avoid making Hypercompressed Mixes

Hypercompression is a form of sound squashing, where everything has an unrelenting and fatiguing intensity, with lost transients and reduced definition. When overused, mastering tools can produce this result, though the tools to do it have migrated to the mixing studio, with a lot of unfortunate sonic results (and a few sonic gems), in my opinion. Hypercompression produces the reverse effect from the intent of a good mix—*boring, lifeless mush*. Perhaps the current slack in music sales is related to hypercompression and its tendency to give everything a monotonous sameness—is the public voting against compression with its pocketbook? Lately it seems about the only place we can enjoy good dynamic range and impact is in the motion picture theatre. This book is partly about how we can bring similar life to our music masters. In this chapter we concentrate on some advice for the mixing engineer.

Let me tell you a sad story. A pop-rock band once sent me a mix that they felt a bit uneasy about, though they could not exactly express why. When I received the DAT it was obvious why. Here's what I heard:

- there was absolutely no dynamic range left, it was "maxxed to the max."
- there was no transient information.
- the sound was grainy and literally lifeless (squashed)
- all the songs sounded continuously and fatiguingly loud. I couldn't listen for more than a couple of minutes at a time.
- although the obvious intent was to produce a hot, clear, punchy sound, the result was exactly the opposite.

No wonder the band felt uneasy, but still they couldn't put their finger on the problem. All the mix elements were there, and the tonality seemed

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* In correspondence. A variation of this quote is in Owsinsky, Bobby. *Mastering Engineer's Handbook*.

† The expressive term **hypercompression** was coined by Lynn Fuston of 3D Audio.

fine. It was easy for me to tell: their engineer had **mixed directly from multitrack through a 3-band mastering compressor to DAT**. In a way I admired his work because he obviously had slaved for hours at the dials "perfecting" this most disappointing sound. Amazingly there were no intermodulation artifacts between the frequency bands, an example of the power of this box, for I was instantly able to identify the brand and type of processor he had used. I called the group and asked them to check if he had made an unprocessed mix as well. Unfortunately he had not. Sadly, I was unable to do anything to salvage this production. I tried a bit of upward expansion (to undo the damage), and the band felt it was an improvement, but an upward expander can only accomplish something when there is "movement" in the source to grab onto (to amplify). Why do you suppose he did this? The motivation was eventually traced to a misguided desire to make the recording "radio-ready" (see sidebar).

Here are some ways to avoid hypercompression during mixing, which easily occurs when consoles and DAWs have a compressor on every channel strip. Everyone has his own style of working with compressors and there are no rules. But I suggest that when learning or beginning a mix, start by working *without* any compressors! Then you'll discover the necessity which was the mother of its invention. The compressor will then become for you a tool to handle problems which cannot be handled with fader moves, not a crutch or substitute for good recording and mixing techniques. Learn about the natural dynamics and impact of musical instruments, then begin to alter them with compressors (which can include using compression to create special effects). Every 5 years or so, give yourself a reality check...try making a recording or mix with little or no compression. You'll rediscover the parts of music that make it lively and aid in its clarity. It's a real challenge, but a refresher course may point out that *less compression* will buy you a more open, more musical sound than you've previously been getting.

The Real Recipe for Radio-Ready

The real recipe for Radio-Ready includes:

1) Write a great original song, use fabulous singers and wonderful arrangements.
2) Be innovative, not imitative.
3) Make sure the music sounds good at home. Keep the dynamics lively, interesting and unsquashed, and some of that virtue will make it through the radio processing.

Start mixing fresh each time—free yourself of preconceptions. Although you compressed the bass on 9 out of the last 10 albums, maybe this time you won't need a compressor. Each musician is an individual and their sound must be respected. In general, the better the bass player, the less compression will be needed, and the greater the chance that compression will "choke up" his sound. If you get to know the sound of your instrumentalists you can then ask yourself: are you trying to capture the sound of your instrumentalists or intentionally creating a new sound? Get a great mix that sounds **alive** and **clear** and **big*** and then later see how much better it can be made in the mastering suite, for mixing and mastering are two different things. After mixing for a while, compare the mix to the raw, unaltered monitor mix (which can be a sobering experience); be honest, have you lost some of the magic that you captured on the recording day? Has the sound closed down instead of opening up?

* Not every piece of music should be *big-sounding*, but I think you get the idea.

The process of refining a mix should always include revisiting your compression (and EQ) settings and questioning your work. Compressors are often used to create a tighter band sound, making the rhythm instruments sit in a good, constant place in the mix. But the wrong compression setting can take away the sense of natural breathing and openness that makes music swing and sway. Thus, I recommend that during mixing, after you've inserted a few compressors on certain instruments (e.g., the bass, rhythm guitar, vocal) and listened for a while, try comparing with the compressors bypassed (total automation makes that process easy; store two fader snapshots so you can switch between them). If you've lost some of the swing, or the subtleties of the musician's performance, then try reducing or eliminating some compression.

I think some of today's mix engineers have to learn (or relearn) the ability to mix loudly and clearly. Rock and Roll music is often a casualty of compressor abuse. I receive rock mixes from well-meaning engineers that should be getting louder and louder and reach a climax, but which have lost their intensity, producing *wimpy loud sound.** There is dynamic inversion; instead of a chorus sounding lively and dramatic, it's been pulled back. To make a better sound and ease the mastering engineer's job, check the climaxes; do they sound open, or squashed? Squashing is a common problem in rock mixes, for it is very difficult to maintain excitement all the way to the highest peaks, but squashing is very hard to repair in mastering. One trick is to start mixing during the climax of the song, make the climax sing and swing, using just enough compression on individual instruments to do the trick; then, return to the beginning, work your butt off riding faders where necessary during the soft passages **but without changing the thresholds from the position used for the peak of the song**. This helps avoid overcompression on the loud passages and keeps the song sounding exciting. It's better to send material that's mixed well and powerfully at the mid levels but at the high levels is not squashed. Even if the climaxes don't sound loud enough to the mix engineer, he should consider it a *work in progress*, for the mastering engineer can take it to the next level of performance, with the punch it needs at mid levels and strength and volume at high levels.

I advise against mix engineers trying to mix through dedicated mastering processors unless you have the patience to refine the many parameters against the constantly-changing parameters of a mix in progress. Even bus compressors built into consoles are not usually optimized for processing overall music. A processor on the bus will change the mix in mysterious ways; it's not predictable whether the vocal or any instrument will stand out, and it can fight the mix instead of helping it. Wideband bus compression causes all the instruments to be modulated by the attack and transients of the loudest instrument. A rim shot or cymbal crash can take down the reverberation and the sound of all the other instruments. Any compressor on a mix bus can quickly become a

* "It's like there has been an unlearning curve. As flexibility has improved, respect for the integrity of the source has all but vanished as people become lost in the possibilities." Bob Olhsson, Mastering Engineer's Webboard.

crutch, a substitute for good mixing techniques. Some mix engineers add delicate bus compression *after the mix has been achieved*, to see if it fattens the sound without deterioration. And to keep the bus compressor from punching "holes" in your mix, they use a very slow attack/release and very little compression (e.g. 1 dB).

Hedge Your Bets. Many mix engineers will subvert **Murphy's Law of Experience** and print two versions to send to mastering, one with bus compression and one without. I often find the bus-compressed version has fatter bass (which the client likes) but wimpy highs and attacks (which the client doesn't like), but in mastering you can have your cake and eat it too: I can supply dynamics processing with carefully-applied multiple time constants, yielding a more impacting result that still has "fat bass." Of course, if the mix was made so aggressively through the bus compressor that removing it would change the mix, then there is no point in providing two versions; be aware that you are painting yourself into a corner, if a remix is not an option.

But what if you want to mix *aggressively*...

This should be the province of the experienced mixer who knows that this is the practice that works best for the particular music, client, or audience and who recognizes the fine subjective line between **aggressive bus** compression and hypercompression. In other words, some engineers mix aggressively on purpose with the bus compressor (or against it); which is only ok if:

- the music truly calls for it
- the experienced mix engineer is aware of all the effects of the bus compressor on the sound

But be careful how you make it loud, because if you deteriorate the clarity of the sound, there's little that can be done to fix it in the mastering. When mixing with aggressive bus compression, I advise you to ascertain the mastering engineer's opinion on this mix in progress. Recently I asked a client why he was using bus compression on his mix, and he replied, "because I think it doesn't sound loud enough without it." But through demonstration, we found out that his mix sounded *wimpy loud* **but not better** (e.g., fatter, punchier, clearer, fuller). I suggest that you concentrate on mixing and save the question of absolute loudness for the mastering; when mixing, go for **better** when auditioned at the same loudness (i.e. turn up the monitor gain until it sounds loud enough). I think Mastering engineers can do a better job and for much music would prefer not to receive bus-compressed mixes—we can stand back objectively, fine-tuning time constants and bandwidths, maximizing the sound quality (and level) without destroying the rhythm, melody or dynamics of the music. Each tune will be optimally and precisely adjusted in the context of the whole album. Attempting these sorts of decisions during mixing, without having the perspective of the entire album, is dangerous since it's irreversible.

> *Learning from your mistakes gives you room to make even bigger ones!*
> — Murphy's Law of Experience

If you wish to try your hand at mastering processing after mixing, by all means do so, perhaps as an example of the type of sound you are looking for, but also bring an unprocessed mix safety to the mastering session.

Monitor gain* has a tremendous effect on these matters of judgment. The higher you place the monitor gain, the less the chance of over-compressing. If the music mix sounds properly "punchy" at a higher monitor gain, then leave the rest of the magic for the mastering rather than add another DSP process or take the sound downhill. The VU meter (as opposed to the peak meter) is our friend. Have one hanging around, preferably calibrated to 0 on the VU meter = -20 dBFS on the peak meter with a sine wave, or if necessary, to as high as –14 dBFS peak. If the VU meter is reading hot, then the sound may be overcompressed.

Stop Emulating Squashed CDs

Many mixing engineers compare their mixes against already-pressed CDs, but be careful what you choose as a standard. Ironically, mastered CDs often do not sound like what comes out of the mix, so how can you emulate something which can only be done post-mix? And emulating aggressively-mastered CDs for a mix may contribute to the vicious circle of escalating loudness. What you really need is to hear the sound of a good mix *before* it was sent for mastering. But since that's not available, choose from the plenitude of pop records that have been well-mixed and conservatively mastered. Visit www.digido.com for *The Honor Roll*, a listing of well-mixed and conservatively-mastered current CDs.

Avoiding Compression Problems during Tracking

When tracking vocalists (who have a habit of belting now and then), a well-adjusted compressor can sound reasonably transparent, and most engineers agree the cure is better than the disease. But watch out for a *closed-in* sound, clamping down when the vocalist gets loud (which reduces clarity and impact), which can be caused by improper time constants, too high a ratio, or using the wrong compressor. Compare IN versus BYPASS before committing to tape. Match levels to make a fair comparison. If you notice too much degradation, maybe it's time to consider a different compressor or change the settings you are using. The sound should be open and clear... remember that no amount of equalization in the mixdown can substitute for capturing a clear sound quality during tracking. This is true for all the lead instruments, including trumpets and electric guitars. If possible, put the uncompressed sound on a spare track—it may save your life. If there's any rule, nine out of ten engineers would prefer to save the decision on drum and percussion compression until mixing. There are always exceptions—every piece of music is unique.

* I prefer the term **monitor gain** to **volume control**. See Chapter 14