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Dynamics Processing Terms and Tips

Audio signals have very wide peak-to-average signal-level ratios, sometimes referred to as *dynamic range*, which is the difference between the loudest level and the softest level. A peak signal can cause overload in the audio- recording or sound-reinforcement chain, resulting in signal distortion. *Dynamics processing* is the process of altering the dynamic range of a signal, thereby enhancing the ability of a live sound system or recording device to handle the signal without distortion or noise and aiding in placing the signal in the overall mix. Common types of dynamics processors include compressors, limiters, expanders, and gates.

To help you better understand this technology, we've provided a collection of frequently asked questions and answers; glossaries of compressor/limiter, expander, and gate terms; and a few suggested compression settings to get you started.

Frequently Asked Questions

What is dynamic range?

Dynamic range can be defined as the ratio between the loudest possible audio level and the lowest possible level. For example, if a processor states that the maximum input level before distortion is +24 dBu, and the output noise floor is -92 dBu, then the processor has a total dynamic range of 24 + 92 = 116 dB.

The average dynamic range of an orchestral performance can range from -50 dBu to +10 dBu, on average. This equates to a 60 dB dynamic range. Although 60 dB may not appear to be a large dynamic range, do the math, and you'll discover that +10 dBu is 1,000 times louder than -50 dBu!

Rock music, on the other hand, has a much smaller dynamic range: typically -10 dBu to +10 dBu, or 20 dB. This makes mixing the various signals of a rock performance a much more tedious task.

Why use compression?

Compressors are commonly used for many audio applications. For example, a vocal performance usually has a wide dynamic range. Transients (normally the loudest portions of the signal) can be far outside the average level of the vocal signal. Because the level can change continuously and dramatically, it is extremely difficult to ride the level with a console fader. A compressor/limiter automatically controls gain without altering the subtleties of the performance. Furthermore, many vocalists move around in front of the microphone while performing, making the output level vary up and down unnaturally. A compressor can be applied to the signal to help correct this problem by reducing the louder passages enough to be compatible with the overall performance.

Let's say that you are mixing a rock performance with an average dynamic range of 20 dB. You wish to add an uncompressed vocal to the mix. The average dynamic range of an uncompressed vocal is around 40 dB. In other words, a vocal performance can go from -30 dBu to +10 dBu. The passages that are +10 dBu and higher will be heard over the mix. However, the passages that are at -30 dBu and below will never be heard over the roar of the rest of the mix. A compressor can be used in this situation to reduce (compress) the dynamic range of the vocal to around 10 dB. The vocal can now be placed at around +5 dBu. At this level, the dynamic range of the vocal is from 0 dBu to +10 dBu. The lower level phrases will now be well above the lower level of the mix, and louder phrases will not overpower the mix, allowing the vocal to "sit in the track."

The same points can be made about any instrument in the mix. Each instrument has its place, and a good compressor can assist the engineer in the overall blend. For example, a kick

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drum can get lost in a wall of electric guitars. No matter how much the level is increased, the kick drum stays lost in the "mud." A touch of compression can tighten up that kick-drum sound, allowing it to punch through without cranking the level way up. A solo guitar can seem to be masked by the rhythm guitars. Compression can make your lead soar above the track without shoving the fader through the roof. Bass guitar can be difficult to record. A consistent level with good attack can be achieved with proper compression, giving the bass the punch it needs to drive the bottom of the mix.

Does every instrument need compression?

Most instruments need some form of compression, often very subtle, to be properly heard in a mix. Of course, if an instrument sits perfectly in a mix without compression, then don't mess with it. But most of the time you'll need it.

What about overcompression?

A well-designed and properly adjusted compressor should not be audible. So if, by "overcompression," you mean that you can hear the compressor working, that's likely to be the result of an improper adjustment on a particular instrument—unless, of course, it is done intentionally for effect.

That said, too much compression can destroy the acoustic dynamic response of a performance. Furthermore, some people use a lot of compression to make their mix as loud as possible to emulate what they hear on the radio, but the top pros usually get that effect by careful compression during mastering, not during tracking and mixing. If you use a lot of compression in your tracks and mix, you might reduce the dynamic range to the point that you haven't left the mastering engineer enough flexibility. So use compression often but don't go over the top unless you are doing it as an effect.

Why use noise gates?

Consider the compressed-vocal example we gave earlier; you now have a 20 dB dynamic range for the vocal channel. Problems arise when noise or instruments (air conditioner, loud drummer, etc.) in the background of the vocal mic become more audible after the lower end of the dynamic range is raised. You might attempt to mute the vocal between phrases in an attempt to remove the unwanted sounds; however this would probably end disastrously. A better method is to use a noise gate. The noise-gate threshold could be set at the bottom of the dynamic range of the vocal, say -10 dBu, such that the gate would shut out the unwanted signals between the phrases.

If you have ever mixed live sound, you know the problems cymbals can create by bleeding through the tom mics. As soon as you add some highs to get some snap out of the tom, the cymbals come crashing through, sending the horn drivers into a small orbit. Gating those tom mics so that the cymbals no longer ring through them will give you an enormous boost in cleaning up the overall mix.

What's the difference between a noise gate and an expander?

The major difference between expansion and noise gating is that expansion is dependent on the signal level after the level crosses the threshold, whereas a true noise gate works independently of a signal's level beyond the threshold.

Which PreSonus products include dynamics processing?

PreSonus makes several hardware dynamics processors, including the ACP88 8-channel compressor/limiter/gate (see **Fig. 1**); COMP16 1-channel compressor; Eureka™ 1-channel preamp/compressor/EQ; Studio Channel vacuum tube channel strip; and StudioLive™ 16.4.2 and StudioLive™ 24.4.2 digital mixers.

FIG. 1: The PreSonus ACP88 8-channel analog compressor offers all of the core features, including stereo linking, Auto Attack, selectable hard/soft knee, and separate sidechain inserts for each compressor and gate.

PreSonus' Studio One™ DAW software includes a variety of dynamics-processing plug-ins, including Channel Strip, Compressor, Expander, Gate, Limiter, Multiband Dynamics, and Tricomp.

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Compressor/Limiter Terminology

Attack. Attack sets the speed at which the compressor acts on the input signal. A slow attack time allows the beginning envelope of a signal (commonly referred to as the initial transient) to pass through the compressor unprocessed, whereas a fast attack time immediately subjects the signal to the ratio and threshold settings of the compressor.

Auto mode. Places a compressor in automatic attack and release mode. The attack and release knobs become inoperative and preprogrammed attack and release curves are used. Current PreSonus hardware products that offer defeatable Auto mode include the ACP88, Studio Channel, StudioLive™ 16.4.2, and StudioLive™ 24.4.2. The COMP16 is entirely preset-based so it essentially is always in Auto mode; however, unlike a traditional Auto mode, the attack an d release settings change depending on which preset is chosen.

Compression ratio. The compression ratio is the relationship between the output level and the input level. In other words, the ratio sets the compression slope. For example, if you have the ratio set to 2:1, any signal levels above the threshold setting will be compressed such that for every 1 dB of level increase into the compressor, the output will only increase 0.5 dB. This produces a compression gain reduction of 0.5 dB/ dB. As you increase the ratio, the compressor gradually becomes a limiter.

Compression. Compression reduces the amount by which a signal's output level can increase relative to the input level. It is useful for lowering the dynamic range of an instrument or vocal, making it easier to record without distorting the recorder. It also assists in the mixing process by reducing the amount of level changes needed for a particular instrument. How severely the compressor reduces the signal is determined by the compression ratio and compression threshold. A ratio of 2:1 or less is considered mild compression, reducing the output by a factor of two for signals that exceed the compression threshold. Ratios above 10:1 are considered hard limiting. As the compression threshold is lowered, more of the input signal is compressed (assuming a nominal input-signal level). Take care not to overcompress a signal, unless you are doing it as an effect, because too much compression destroys the dynamic response of a performance.

Compressor. A compressor is a type of amplifier in which gain is dependent on the signal level passing through it. You can set the maximum level a compressor allows to pass through, thereby causing automatic gain reduction above some predetermined signal level, or threshold. Punch, apparent loudness, and presence are just three of the many terms used to describe the effects of compression.

Frequency-conscious compression. In *frequency-conscious* or *frequency-dependant* compression, a full-band compressor acts on the entire signal but the detector is set to be triggered by the presence of specific, user-selected frequencies.

Hard/soft knee. With hard-knee compression, the gain reduction applied to the signal occurs as soon as the signal exceeds the level set by the threshold. With soft-knee compression, the onset of gain reduction occurs gradually after the signal has exceeded the threshold, producing a more musical response (to some folks). The term "knee" refers to the way the compression curve bends at the threshold point when represented graphically (see **Fig. 2**). All current PreSonus hardware compressors have hard/soft knee switches except for the COMP16, which is entirely preset-based.

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FIG. 2: The term "knee" refers to the way the compression curve bends at the threshold point when represented. This curve shows hard-knee compression.

Multiband/split-band compression. A multiband or "split-band" compressor divides the audio signal into two or more frequency bands so each band can be compressed independently. That lets you compress, for example, a guitar's bass frequencies differently from the high frequencies. For examples, check out the PreSonus Multiband Dynamics plug-in for Studio One, which provides five independent compression/expansion bands.

Limiter. At the simplest level, a limiter is a compressor that is set to prevent any increase in the level of a signal above the threshold. For example, if you have the threshold knob set at 0 dB, and the ratio turned fully clockwise, the compressor becomes a limiter at 0 dB, so that the output signal cannot exceed 0 dB regardless of the level of the input signal. Typically, compression ratios of 10:1 are considered to be limiting.

A true analog peak limiter is not just a compressor with a high ratio. A compressor's detector circuit is usually designed to detect RMS, or average, levels, so transient peaks will usually overshoot a compressor's threshold level. A true peak limiter employs a detector circuit that responds to peak energy levels and thus reacts faster.

All current PreSonus hardware compressors can be used for limiting, albeit they are not true peak limiters in the sense described here.

Look-ahead compression. With digital compressors, such as the PreSonus Channel Strip, Compressor, Multiband Dynamics, and Tricomp plug-ins for Studio One, the compressor can analyze what it is about to process ("look ahead") and can place the attack time right at the onset of—or even before—the sound, resulting in zero attack time. This is great for catching unwanted transients but it must be used with care so as not to remove desirable transients, such as the attack of a snare drum.

Makeup gain. When compressing a signal, gain reduction usually results in an overall reduction of level. The gain control allows you to restore the loss in level due to compression (like readjusting the volume).

Release. Release sets the length of time the compressor takes to return the gain reduction back to zero (no gain reduction) after the signal level drops below the compression threshold. Very short release times can produce a very choppy or "jittery" sound, especially in low-frequency instruments such as bass guitar. Very long release times can result in an extremely compressed sound; this is sometimes referred to as "squashing" the sound. All ranges of release can be useful at different times, however, and you should experiment to become familiar with the different sonic possibilities.

Sidechain. Compressors have a sidechain detector circuit that is not in the audio path but "sees" when the threshold has been exceeded and tells the compressor's gain-control element when to attenuate the audible signal. The circuits for threshold, ratio, attack, and release are found in the sidechain.

Many compressors provide sidechain inserts on their rear panels that interrupt the signal that the compressor is using to determine the amount of gain reduction it should apply. When no connector is inserted into this jack, the input signal goes directly to the compressor's control circuitry. When a connector is inserted into this jack, the signal path is interrupted, enabling you to send the control signal to an outboard processor, such as an equalizer, and return the processed signal before the threshold detector. This is commonly used to reduce sibilance (de-essing) in a vocal track.

You also can use the sidechain insert for *ducking*: reducing the level of music or other background sound whenever a narrator speaks or vocalist sings. You've often heard the effect of this in broadcasting, where ducking ensures the announcer's voice is heard. In this application, the vocal signal is routed to the sidechain input, while the music is routed through the main compression circuitry. Now the compressor will automatically *duck*—that is, reduce the level of—the music whenever the narrator speaks or the vocalist sings.

The ACP88 is the only current PreSonus hardware compressor with a sidechain insert but all Studio One compressor plug-ins offer this feature.

Stereo linking. When compressing a stereo instrument or mix, you might want to compress both channels equally so that the levels are attenuated by the same amount. That way, one side's level won't drop more than the other, messing up the stereo image. You can accomplish this by linking the two channels of a dual-channel compressor or by linking adjacent channels of a multi-channel compressor such as the PreSonus ACP88. In some forms of stereo linking, the channel that exhibits the most gain reduction determines the gain reduction for the other channel.

Another form of stereo linking, found in the ACP88 and in the StudioLive-series digital mixers' Fat Channel compressors, establishes a master/slave relationship between the two channels in which one side (typically the left) is the master; the slave channel follows the master's compression pattern.

Threshold. The compressor threshold sets the level at which compression begins. When the signal is above the threshold setting, it becomes eligible for compression. Basically, as you turn the threshold knob counterclockwise, more of the input signal becomes compressed, assuming you have a ratio setting greater than 1:1.

Expander Terminology

Downward expansion. Downward expansion is the most common expansion used in live sound and recording. In contrast to compression, which decreases the level of a signal after it rises above the compression threshold, expansion decreases the level of a signal after the signal goes below the expansion threshold. The amount of level reduction is determined by the expansion ratio. For example, a 2:1 expansion ratio reduces the level of a signal by a factor of two. (e.g., if a level drops 5 dB below the expansion threshold, the expander will reduce it to 10 dB below the threshold.)

Commonly used for noise reduction, expansion is very effective as a simple noise gate. The major difference between expansion and noise gating is that expansion is dependent on the signal level after the level crosses the threshold, whereas a true noise gate works independently of a signal's level beyond the threshold.

This type of expansion reduces the level of a signal when the signal falls below a set threshold level. This is most common used for noise reduction.

Dynamic expansion. This is basically the opposite of compression. In fact, broadcasters use dynamic expansion to "undo" compression before transmitting the audio signal. This is commonly referred to as *companding* or COMPression followed by expANDING.

FIG. 3: This full-featured Expander plug-in includes a sidechain with Key Filter and Key Listen.

Expander. Expanders increase the dynamic range of a signal after the signal crosses a threshold (**see Fig. 3**). There are two basic types of expansion: *dynamic* and *downward*.

Expansion ratio. The expansion ratio sets the amount of reduction applied to a signal once the signal has dropped below the expansion threshold. For example, a 2:1 expansion ratio attenuates a signal 2 dB for every 1 dB it drops below the threshold. Ratios of 4:1 and higher act much like a noise gate but without the ability to tailor the attack, hold, and release times.

Gate Terminology

Attack. The gate attack time sets the rate at which the gate opens. A fast attack rate is crucial for percussive instruments, whereas signals such as vocals and bass guitar require a slower attack (see **Fig. 4**). Too fast of an attack can, on these slow-rising signals, cause an artifact in the signal, which is heard as a click. All gates have the ability to click when opening but a properly set gate will never click.

FIG. : Gating a vocal with Studio One's Gate plug-in. Since vocals are slow-rising signals, in terms of dynamics, we're using a slow attack time.

Hold. Hold time is used to keep the gate open for a fixed period after the signal drops below the gate threshold. This can be very useful for effects such as gated snare, where the gate remains open after the snare hit for the duration of the hold time, then abruptly closes.

Key Filter. The Key Filter is a filter with a variable frequency that allows you to remove problem frequencies from the gate's trigger signal. Let's say you want to gate the snare drum so that the gate opens every time the drummer hits the snare, letting the snare through, and then closes again. That can be used as an effect or simply so that the noise leaking into the snare mic won't muddy up the overall sound. To make the gate open, you use a copy of the dry signal from the snare mic to trigger the gate, allowing the snare sound to pass. The problem is, other instruments – notably cymbals – often leak into the snare mic, and they could accidentally trigger the gate. The key filter enables you to filter out the high frequencies in the cymbals so that only the snare signal remains to trigger the gate.

Key Listen. Key Listen lets you audition the gate's trigger signal so you can adjust the Key Filter's frequency control until the problem frequencies are eliminated.

Noise Gating. Noise gating is the process of removing unwanted sounds from a signal by attenuating all signals below a set threshold. As described, the gate works independently of the audio signal after being "triggered" by the signal crossing the gate threshold. The gate will remain open as long as the signal is above the threshold. How fast the gate opens to let the "good" signal through is determined by the attack time. How long the gate stays open after the signal has gone below the threshold is determined by the hold time. How fast the gate closes is determined by the release. How much the gate attenuates the unwanted signal while closed is determined by the range.

Noise gates were originally designed to help eliminate extraneous noise and unwanted artifacts from a recording, such as hiss, rumble, or transients from other instruments in the room. Since hiss and noise are not as loud as the instrument being recorded, a properly set gate will only allow the intended sound to pass through; the volume of everything else is lowered. Not only will this strip away unwanted artifacts like hiss, it will add definition and clarity to the desired sound. This is a very popular application for noise gates, especially percussion instruments, as it will add punch or "tighten" the percussive sound and make it more pronounced.

Range. The gate range is the amount of gain reduction that the gate produces. Therefore, if the range is set at 0 dB, there will be no change in the signal as it crosses the threshold. If the range is set to -60 dB, the signal will be gated (reduced) by 60 dB.

Release. The gate-release time determines the rate at which the gate closes. Release times should typically be set so that the natural decay of the instrument or vocal being gated is not affected. Shorter release times help to clean up the noise in a signal but may cause "chattering" in percussive instruments. Longer release times usually eliminate "chattering" and should be set by listening carefully for the most natural release of the signal.

Threshold. The gate threshold sets the level at which the gate opens. Essentially, all signals above the threshold setting are passed through unaffected, whereas signals below the threshold setting are reduced in level by the amount set by the range control. If the threshold is set fully counterclockwise, the gate is turned off (always open), allowing all signals to pass through unaffected.

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Compression Settings: Starting Points

The following are the compression presets that were used in the popular but discontinued PreSonus BlueMax. We have included them as a jumping-off point to get you started.

Vocals

Soft. This is an easy compression with a low ratio setting for ballads, allowing a wider dynamic range. It's good for live use. This setting helps the vocal "sit in the track."

Threshhold	Ratio	Attack	Release
-8.2 dB	1.8:1	0.002 ms	38 ms

Medium. This setting has more limiting than the Soft compression setting, producing a narrower dynamic range. It moves the vocal more up front in the mix.

Threshhold	Ratio	Attack	Release
-3.3 dB	2.8:1	0.002 ms	38 ms

Screamer. This setting is for loud vocals. It is a fairly hard compression setting for a vocalist who is on and off the microphone a lot. It puts the voice "in your face.

Threshhold	Ratio	Attack	Release
1.1 dB	3.8:1	0.002 ms	38 ms

Percussion

Snare/Kick. This setting allows the first transient through and compresses the rest of the signal, giving a hard "snap" up front and a longer release.

Threshhold	Ratio	Attack	Release
-2.1 dB	3.5:1	78 ms	300 ms

Left/Right (Stereo) Overheads. The low ratio and threshold in this setting gives a "fat" contour to even out the sound from overhead drum mics. Low end is increased, and the overall sound is more present and less ambient. You get more "boom" and less "room.

Threshhold	Ratio	Attack	Release
-13.7 dB	1.3:1	27 ms	128 ms

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Fretted Instruments

Electric Bass. The fast attack and slow release in this setting will tighten up the electric bass and give you control for a more consistent level.

Threshhold	Ratio	Attack	Release
-4.4 dB	2.6:1	45.7 ms	189 ms

Acoustic Guitar. This setting accentuates the attack of the acoustic guitar and helps maintain an even signal level, keeping the acoustic guitar from disappearing in the track.

Threshhold	Ratio	Attack	Release
-6.3 dB	3.4:1	188 ms	400 ms

Electric Guitar. This is a setting for "crunch" electric rhythm guitar. A slow attack helps to get the electric rhythm guitar "up close and personal" and gives punch to your crunch.

Threshhold	Ratio	Attack	Release
-0.1 dB	2.4:1	26 ms	193 ms

Keyboards

Piano. This is a special setting for an even level across the keyboard. It is designed to help even up the top and bottom of an acoustic piano. In other words, it helps the left hand to be heard along with the right hand.

Threshhold	Ratio	Attack	Release
-10.8 dB	1.9:1	108 ms	112 ms

Synth. The fast attack and release on this setting can be used for synthesizer horn stabs or for bass lines played on a synthesizer.

Threshhold	Ratio	Attack	Release
-11.9 dB	1.8:1	0.002 ms	85 ms

Orchestral. Use this setting for string pads and other types of synthesized orchestra parts. It will decrease the overall dynamic range for easier placement in the mix.

Threshhold	Ratio	Attack	Release
3.3 dB	2.5:1	1.8 ms	50 ms

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Stereo Mix

Stereo Limiter. Just as the name implies, this is a hard limiter, or "brickwall," setting—ideal for controlling the level to a two-track mixdown deck or stereo output.

Threshhold	Ratio	Attack	Release
5.5 dB	7.1:1	0.001 ms	98 ms

Contour. This setting fattens up the main mix.

Threshhold	Ratio	Attack	Release
-13.4 dB	1.2:1	0.002 ms	182 ms

Effects

Squeeze. This is dynamic compression for solo work, especially electric guitar. It gives you that glassy "Tele/Strat" sound. It is a true classic.

Threshhold	Ratio	Attack	Release
-4.6 dB	2.4:1	7.2 ms	93 ms

Pump. This is a setting for making the compressor "pump" in a desirable way. This effect is good for snare drums to increase the length of the transient by bringing the signal up after the initial spike.

Threshhold	Ratio	Attack	Release
0 dB	1.9:1	1 ms	0.001 ms

FOR MORE READING:

"The Big Squeeze," *Electronic Musician*, February 2001
"Let's Split," *Electronic Musician*, January 2004
"Multiband Dynamics," *Recording*
"Compression Made Easy," *Sound on Sound*, September 2009