POWER USER

Motif XS:
VCM Virtual Circuit Modeling:
Understanding Compressor/Limiters

The VCM Compressor 376

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Compressor/Limiter: VCM Compressor 376

Settings on Compressor/Limiters are very much all about the current input. In other words, you cannot make the settings for Compressor/Limiters without listening to your current input source. There are several things that can improve your understanding and use of the Compressor/Limiter:

First thing is to understand that the device is in the *amplifier* family. It is technically speaking a "Leveling Amplifier" because it helps the recording engineer control erratic levels during record or mixdown. As a human you are not physically fast enough to "ride the faders" and react to a signal that is heading for the red zone, above 0VU. Therefore, these leveling amplifiers are used to automatically detect the strength (voltage) of a signal and apply gain reduction. You can think of a compressor/limiter as a really fast set of hands – It is quick enough to react to level that would distort – and to lower that level before distortion occurs. It is able to bring the level down prior to bad things happening. They help the engineer control the **Dynamic Range** of the incoming signal…

The second concept to understand is this Dynamic Range function. When setting record level, the engineer will ask the musician for a run through. This is so they can determine the Dynamic Range of the performer. That is, 'how loud' **and** 'how soft' a signal they will be creating. This is necessary because they cannot allow the musician to overload (distort) the system. Without a compressor/limiter, the 'how loud' they get function will determine the Input Gain setting… as they will simply set the maximum amount of level without the VU meter going into the red. The softest sounds will, of course, fall below this and will fall wherever they may.

A problem arises when you start to add other instruments. The soft nuance of a particular instrument can get lost in the overall musical signal. There are certain of these subtle sounds that you want bring forward. You do this by adjusting the Dynamic Range of the input signal.

Dynamic Range is measured in dB, which is a ratio. A Ratio is a comparison of one thing to another. In this case the loudest sound to the softest sound a particular input device is going to make. By compressing or limiting that distance between loud and soft, the leveling amplifier makes signals easier to handle – because as we all know distortion of recording level is a very bad thing. The compressor reduces the gain of the loud, and leaves the soft alone – narrowing the dynamic range.

The difference between compression and limiting is simply a matter of degree.

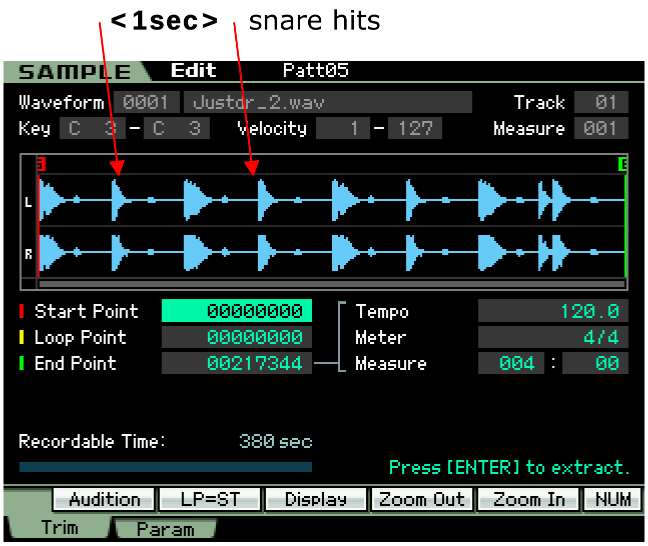
A **compressor** is an amplifier where the *more* signal you put in; the *less* signal you get out. This occurs once you exceed a specific level (threshold). A **limiter** is an amplifier where the *more* signal you put in; the output remains virtually the *same* and does not perceivably increase. This occurs once you exceed a specific level (threshold).

We mentioned ratios… 2:1 (read two to one), 3:1, 4:1 and 8:1 are considered compression ratios, while 10:1 and above are considered limiter ratios.

The third thing to understand (and a very important thing) is you do not want to put a compressor on everything individually – you will be back to where you started and you will kill the life of your music. Compressors are used to make things like vocals stand out in the mix. If soft words or syllables are getting lost in the mix, a compressor can allow that signal to be brought forward in the mix. You can give a sound more ***presence***. However, if you do the same thing to every sound, it kind of loses its uniqueness and effectiveness (more on this point in a minute).

UNDERSTANDING THE PARAMETERS:

Attack - This setting means that at least the first few milliseconds (ms) of the signal will pass before the Compressor engages. Snare drums are a rather fast attack with a big *transient* peak at the attack - you might want to make that setting faster (lower the number) for percussive sounds. A transient peak is the stick hitting the skin of the drum. It helps to think of all sounds as a two-part scenario: "cause and affect"; The stick hitting the skin (causing a large transient peak) and then the response of the drum (the body of the sound). Vocals (words and syllables) vary greatly in level and make for a very erratic incoming signal. Certain words are fast attack words, “blue”, “people”, while others can develop less fast “freedom”, “sizzling”. Setting attack is a matter of paying attention to what the results sound like. You cannot just make a setting and assume it is right – do not be afraid to adjust this. But the general rule of thumb is think about whether a sound is a *triggered event* sound with a percussive peak (this includes instruments that are hammered, plucked or struck). Or does the instrument’s sound develop slowly out of silence. This would include blown and bowed instruments. A Clarinet note can develop slowly up from silence – well, more slowly than a snare drum attack. You will notice that Attack is a TIME parameter and therefore is measured in millisecond. One millisecond is one-thousandth of a second. Now that may be meaningless to you as a musician, as we do not think in terms of time like that – a musician’s time is divided by beats per minute, after all. But in general what you are looking for is an attack time setting that allows for the type of input you are working with – for example, an attack setting of 150-200ms for a snare drum is far too slow of an attack. The sound could totally rise and fall within that time. Think about how quickly a snare drum comes and goes. Let’s do some simple math (promise it will not be painful): take a musical tempo of 120 beats per minute. If the time signature is 4/4 time, at this tempo basically two beats in a second. Therefore a snare drum played on the backbeat (beats 2 and 4) will occur exactly a second apart. If you ever look at a sample of a drum loop where the snare is being struck every other beat, you realize that it does not fill the distance between the beats. In fact, there is plenty of room between the 2 stroke and the 4 stroke – you could drive a large truck through that gap! The attack of snare is very rapid in terms of time.



Shown at left: Here you can see a basic Kick-hihat-Snare groove in the XS Sampler (2 measures are shown). The red arrows point out the snare hits on beat 2 and beat 4. At 120BPM that is precisely one second of time. You can see that the entire snare event is just a small slice of time.

If your attack time is not set fast enough the entire sound could “escape” undetected. Not to worry, you set this by ear (and experience) – we just wanted to make sure you are in the ballpark and not missing the entire event because you are unable or unaware of how time in milliseconds relates to musical passages.

Release – The release setting is subjective and will be dependent on your tempo. This is the time it takes for the Compressor/Limter to 'let go' of the signal. Once the device reduces the gain (the VU meter will deflect), *Release* is the time it takes for the indicator to return to 0VU. Because a snare does not really 'sustain' in the sense of like a guitar sustaining, this is very subjective and passes probably without much notice - if you were using this on a musical sound - guitar players use long releases to get a perceived increased sustain. The picked attack is the transient peak and has lots of energy, the sound then dies a bit (called the initial decay) to the sustain level (main body of the sound). If you compress that big peak of energy on the attack the whole signal seems to remain at the same volume longer...and is perceived as longer sustain. The more you compress and the longer the release the longer the perceived sustain. (This describes the use of compressors mostly for long musical sounds with long envelopes). Release is basically the time it takes for the compressor to stop compressing the signal. You cannot and do not really change the physics of how long the signal lasts, what you can do is change the human perception of how it is changing of the time it is happening. You cannot make a snare drum or a guitar note last longer than it physically does, but by changing the loudness "shape" (the envelope) you can make it seem like it stayed the same volume for a longer period of time. If your Release is set too long, the compressor does not let go and the next event will also be compressed. Therefore, think about our kick-hihat-snare groove… if you want the compressor to be available again for the next hit you want to know approximately how much time that has elapse between events.

Input – This parameter determines both the relative *Threshold* and the amount of *Gain Reduction*. Compression/Limiting begins at a point (in this case, called a fixed threshold); Classic analog leveling amplifiers often used this fixed threshold concept – you adjust the INPUT and work with the amount of GR (or Gain Reduction) that is required to make the sound behave. Remember the goal is not to squash the sound completely just reduce the loudest peaks so we can have a more controllable signal strength (level).

> **Threshold** – Is the point at which compression/limiting begins (It's all about the result!) The threshold is a critical setting and must be made with considerable care. On the VCM 376 the INPUT parameter will determine both input **and** the amount of Gain Reduction. The amount of Gain Reduction will be shown on the Meter. As you increase the INPUT of the signal you will notice that the VU METER set to show GR (Gain Reduction) will start to deflect downward. It must be mentioned here that the incoming signal can very often have its own OUTPUT level, which becomes INPUT to the compressor. The downward deflection of the GR meter shows the amount the signal's peak is being reduced.
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> **Gain Reduction –** is a measure of how much in dB (shown on the Meter) you are reducing the dynamic range of the signal. In general, you want to reduce just the very loudest peaks of an incoming signal. So when GR is selected you will see the meter deflect down from 0VU.

The Attack time will be seen as how quickly the indicator move down from 0VU, while the Release time will be seen as how slowly the indicator returns to 0VU

Ratio – A ratio of 10.0:1 (ten point zero-to-one) will make the Compressor work like a LIMITER. The difference between a Compressor and a LIMITER is one of degree (10:1 and above are LIMITER). While settings like 2:1, 3:1, 4:1 are typical settings for compression. A Setting of 10:1 (Limiting) will take a signal level that wants to go to 10dB above the threshold but will only let it get to 1dB hotter (that's the 10-to-1). For each 1dB of signal it allows the signal to increase it would need the energy to go 10dB. That is, if the signal was strong enough to reach the threshold point and want to go 10dB above that point then the compressor only allows it actually increase 1dB above that point. If it would go 20dB above the threshold the LIMITER would only allow it to go to 2dB, and so on.

Output – Output level is so that you can return the now compressed dynamic range to its proper balance within the mix. The idea with a Compressor/Limiter is you are making the soft level closer to the loud level (i.e., compressing the dynamic range) so the output is very important to restore the balance (the level within the whole mix). Think of a singer who whispers over a big band track - whispers are soft. Then the singer belts out at the top of their lungs - that's the loud. A

Compressor would bring the loudest signal down closer to the whisper's level... compressing the dynamic range...then the all important OUTPUT would let you raise the overall signal back up. So the whisper will sit more up-front and be heard in context of the mix yet the shout will not clip the levels! That's the purpose of the Compressor/Limiter. We reduce the distance between soft and loud by controlling how loud something can get, then we raise this now tighter (more present) signal back to the forefront. The 'soft' segments can now compete with the rest of the musical signal without the 'loud' signal going out of control.

Background: We mentioned transient peaks - These are very rapid and very pronounced level excursions at the front end of a sound. A snare drum whack can create transient peaks that are as large as 10-15dB (or more) above the rest of the sound (the response of the drum) and are typically too fast for most VU meters to register. You cannot allow this transient peak to just clip the input – this can cause all kinds of trouble down the line. Compressors are used to reduce that peak so that the energy is not lost. Here's what that means: If you want a nice fat snare drum sound, you will not be able to get the body of the sound loud enough if that pesky *transient peak* forces you to lower the gain on the snare drum track. By reducing the dynamic range (bringing the peak down closer to the body of the sound) we can now raise the overall level of the snare drum input and the body of the sound can have a greater impact before distortion. We call that phenomena ***presence***.

The letter "P" in the word "petunia" is called a *plosive* – because of its explosive transient. You do not want to have to adjust the level of the signal down to accommodate the "Pe" if it means you lose the intelligibility of the rest of the word. A compressor would gently compress the "Pe" and give more focus on the "-tunia" portion of this word.

When you hear engineers say compressors give a signal "more weight" or "more presence" in the mix this is what they are talking about. The body of the sound is not buried way down in level because the transient peak forces them to lower the overall sound. You are cheating the rules of physics. Of course, there is such a thing as "over-compression", and this can be a very bad thing – so these settings must be made with great care.

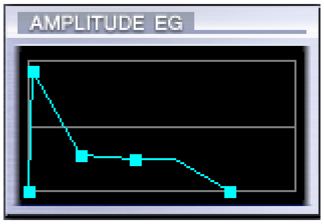
Over-compression is often described as "breathing" or "pumping" – this is due to the phenomena that occurs. Picture a singer in front of microphone; among the soft sounds that they make are lip smacks and drawing breath. These sounds typically go unnoticed by the listener… until you OVER-compress the dynamic range (this is a bad thing). Remember you are bringing the loudest peaks down closer to the softest sounds and then you are bringing the overall output up again. So now things that were heretofore ignored are now right in the listener's face. The singer will sound like they have emphysema as they draw a breath, because the body of what they are saying is compressed so that it is virtually the same volume as the soft sounds. This is described as hearing the compressor "breathe". The "pumping" description comes from an improperly set release time. Release is how quickly or slowly the compressor allows the dynamic range reduction to return to normal. If you become too aware of this change you start feeling like the air was sucked out of the room then it gets blown back up again – as if somebody was pumping in and withdrawing air from the environment. The silence will feel like fresh air and when the voice comes in the air is removed. It becomes physically exhausting to listen to and will make your listeners fatigued – and they will not know why…

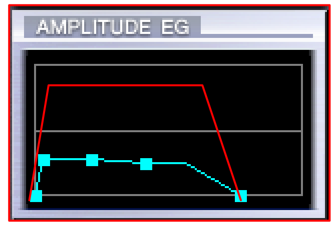
The best rule of thumb for working with compressors and limiters is: *If you notice it, you probably have* ***too*** *much. It is best felt, not heard.* This makes it difficult for newbies, because most people turn knobs full on or full off and make wide changes, this is all about subtlety. And thus is a difficult one to master.

Rules are meant to be broken…

Now all that said, there are guitarist, who have done with compression much the same as they have with distortion. They have thrown the book out and they use compressors strictly as an effect. They use the peak gain reduction to give the *illusion* of longer sustain. By flattening the peak of a waveform it seemingly remains at the same volume for a longer time – and gives the feel that you are getting more sustain. Sustain being a volume level that remains consistent for a time. It is actually an aural illusion.

In the graphic below the vertical axis (up/down) is loudness, and the horizontal axis (left-to-right)





is time moving forward. A typical plucked string amplitude envelope is shown at left with a strong transient peak. While at right you see the same pluck with the transient removed – the signal rises to full volume and remains at that volume giving the illusion of longer sustain… Yes, compression does make the overall signal perceptible softer… it is after all reducing the gain and making the loud sound closer to the soft sounds. But this is why the OUTPUT parameter is provided and is very important. You can then boost the overall output of the compressed dynamic range to restore unity gain (i.e., restore the loudness) shown as the red line; except now it will be a much "heavier", "more present" sound. Notice the envelope of the whole sound does not last any longer really but the perception will be that there is "longer sustain" – because instead of climbing and descending the mountain (transient peak), the level reaches a plateau and remains there.

VCM Compressor 376

Screen shot of the VCM ADD-ON EFFECT for 0-series, DM-series and PM-series mixers on which the Compressor 376 is based.

The geniuses at Yamaha's **K's Lab** in Hamamatsu have created (recreated) the response and behavior of the classic Compressor/Limiters of the past. What VCM (Virtual Circuit Modeling) does is not just mimic the device's results (as is the case with most of the modeling devices out there)… what they have done is model the actual electronic components used in the classic leveling amplifiers. The tubes, transistors, resistors and capacitors used to build these devices, their tolerances; their response to heat variation and how they were used in the actual circuitry of the devices is what has been modeled. This means that the input-output signal flow will behave the same as the original device under similar conditions. By modeling the components and the circuitry they can virtually build almost any device. The unique charm of these classic devices is what has been captured with the VCM technology.

Now if you only notice that the VCM effect processors are *really cool* that is ultimately all the designer's wanted to accomplish. But if you are an "old school" recording engineer like me, it was the response and tone that you could get from the gear that made it so good in the first place. I find myself getting really excited about VCM because of the authentic feel to the sound you get. One of the early complaints about "digital" technology was that it was "too precise" and lacked a personality. Now that is a funny idea to explain. How can something be too precise? Don't we want accuracy? Well in some cases you don't, actually. Old compressors were naturally slow to respond, where digital components can be very quick. Some old compressors did not even have an Attack parameter. It simply responded as fast as it could – and this was actually rather slow. But this also gave some of these old compressors a certain charm (and often a specific use). This slowness made some of them perfect for things like vocals. So you had some devices that had reputations for being essential for certain types of sounds. Others were really inaccurate at the point of engagement (Threshold) – where the signal deflects – and their response was not precise. The concept of a soft knee was born. Rather than a sharp angle of deflection, the response due to the components used in the circuitry had a more rounded response (knee). Later in transistor-based gear this was so endearing it became a programmable feature. So the charm actually became a controllable feature. Now when things first went digital – everything was super precise and a lot of the old circuit charm was lost or you had to work harder to achieve it. This is what K's Lab has found again.

By approaching modeling of these devices on the component level, the modeled device "behaves" exactly like the original. This is what makes the VCM effects so very unique and valuable. Rather than simply modeling one specific device, they have come up with a much more flexible product. Physical Modeling is being used and touted in a lot of products these days, but it should not be an end in itself. Toshi Kunimoto, the developer of this technology (and the 'K' in K's Lab), has been at work on physical modeling since 1987 – he also is responsible for the incredible VL and VP synthesizer technologies. In fact, the VL and VP were offshoots to his research on effect processing. The importance of developing a *musically* useful series of devices was at the core of the research. Modeling the circuitry was the way to get there (he returned to work on studio processors in 2001). Much testing and evaluation on VCM's musicality was done and it is one of the most significant advances in signal processing in a very long time. As with the Physical Modeling synthesizers, "behavior" is a key concept to understand. Because of the construction of the original device it responds a particular way under particular circumstances - by modeling the things that make up the device you can have your model behave the same way as the original device. Compressors can be used to control level (leveling amplifier) as we have discussed, but they can also be used subjectively to increase sustain (as guitar players use them).

Keep in mind that you are increasing the impact, the presence, the "in-your-face" quality of the music without appreciably increasing the overall volume. Sometimes it is hard to believe that the overall volume did not increase – only your perception of volume increases.

"The VCM …can create the nuance only analogue could bring, while enjoying the convenience only digital can provide. It isn't difficult for the VCM to reproduce even the sound of tube amps, stomp boxes, or expensive vintage analogue consoles in digital. It might sound like a dream, but it is not. Technology is already here." - Toshi Kunimoto