Level it Out, and Shut the Gate – MTEC 2013

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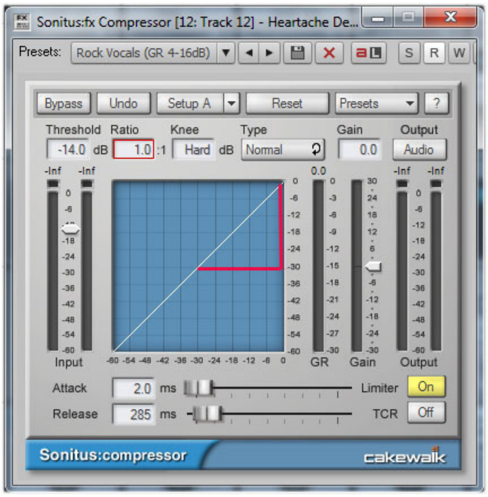
Part 1: Audio Compression

Whether it's in a live performance situation or doing recordings, many of us have spent time trying to even out variations in sound levels of vocalists, instruments, and even the sound levels of a whole mix. We have had our hands almost glued to mixer faders to adjust to the changes.

Audio Compression provides us with the opportunity to adjust the way changes in signal level are dealt with.

When there is no audio compression applied to a signal, whether it be in a live mixing or recording scenario, any changes in level are passed on directly to the destination. This represents a 1 to 1 correspondence between the source input and output to the next stage.

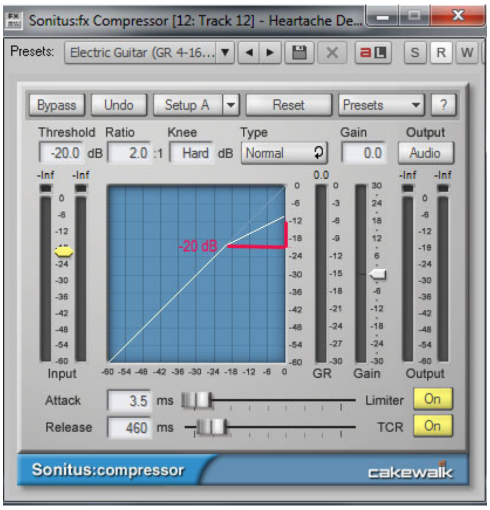


In graphical form, this direct relationship looks like a straight line as shown:

The horizontal axis of the graph shows the incoming signal and the vertical axis shows the outgoing signal.

If the input signal level changed by 30dB for example, the output signal would change by the same amount, as shown in red.

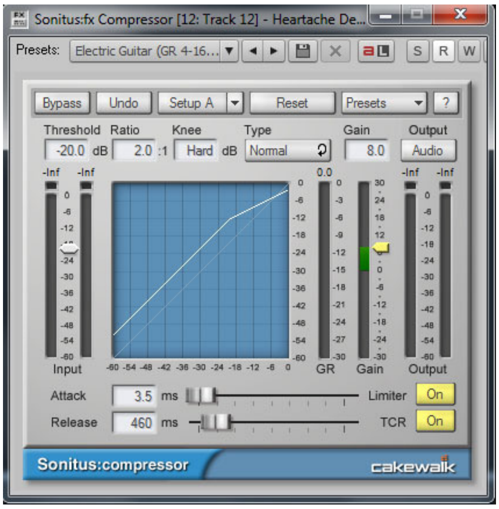
Applying compression above a certain audio level called the threshold, changes that one to one correspondence, and leads to a smaller change in output for a given change in input. In the example shown below, when the signal is below -20dB, there is 1 to 1 change, but when the signal level reaches above the -20 threshold level, every 2 dB increase of signal level produces only a 1db increase in output level.



In this case we have a ratio of 2.0 to 1, and that is shown in the parameters above the top LHS of the graph. This would have the effect of reducing how much the volume of the signal changed by, and limiting the output signal to a maximum of -10 dB.

The reduction in variation is what we were trying to achieve, but it may be desirable to allow the output to reach it close to its maximum point before distortion, which is 0 dB, otherwise the signal may get a little lost in the mix. In order to achieve both, we can use what is called gain compensation, ie adding some signal level to all frequencies.

In the following example around 8dB of gain has been added.

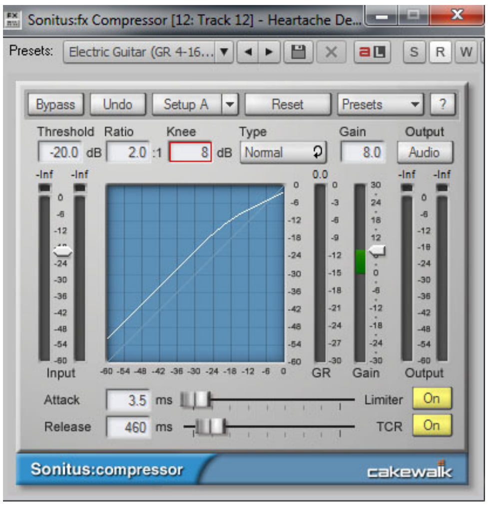


This will have the effect of boosting the signal level when it is soft, but reducing the changes of level when it starts to get loud, and always keeping it below a potential distortion point, particularly in the digital recording world.

In the example shown, the incoming signal level may change from -60dB to 0dB, but the dynamic range of the output has been reduced to a 50dB change.

The term “compression” refers to this squeezing of the dynamic range (the difference between the softest and loudest signal levels).

In order to adjust for a range of input source types, such as lead vocals, guitar, kick drum, or a whole Front of House Mix, there are several other key settings which can be adjusted in a typical audio compressor. These include:



The Knee – This controls the behavior of the compressor around the threshold level. The knee shown above is a hard knee, meaning that the changes applied to signal level are applied fully, once the incoming level exceeds the threshold.

For a more gradual and smooth change, a soft knee can be applied, as shown on the left.

The Attack – This setting adjusts the time taken (in milliseconds (ms)) for the compression to reach its maximum effect when the threshold has been reached. Lengthening this time allows for the detail of the start of the signal to pass through more. To get a good idea of typical settings, try many of the presets you will find either in a recording compressor plug in, or in digital mixing systems. The range is usually from about 1 ms for a snare drum, a few ms for vocals, and up to 30 to 40 ms for mixes.

The Release – this adjusts the time it takes for the compression effect to disappear once the signal level has fallen below the threshold. This is generally much longer than the attack time, and depends on the typically decay of the source sound. Typical values are between 150 ms and up to around 800 ms. Too short a time will result in noticeable changes in signal level as the decaying sound reaches the threshold.

In both the recording and live scenarios delivered by digital mixing and processing systems, remember to save the settings you have made for future instant recall.

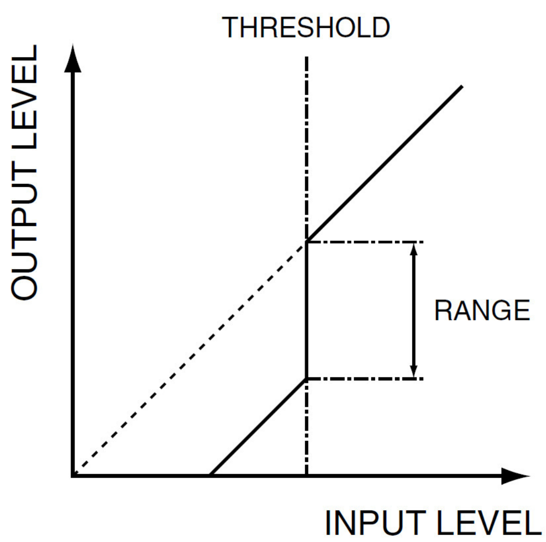
Part 2: Using Gates to Clean up the Mix

The other issue many of us have faced in school performances is the lack of ability to achieve a clean and clear sound for the audience. There are a great many factors which contribute to this, including the nature of the acoustics of the room, the level at which performers may have amplifiers on stage set at, the quality of microphones and instruments being used, to name just a few.

A discussion of how to deal with all of those factors is beyond the scope of this paper, however many modern digital mixing systems provide us with key tool used routinely by sound engineers in concerts.

A significant factor in the muddiness of live sound is the number of mics which need to be used on stage. In school productions, it is not uncommon to have at least a dozen mics onstage, and large systems can easily double that. The reality is that much of the time, the mics are picking up not only the sounds they are placed on stage to detect, but are also picking up all of the sounds on stage to some degree. When there are many of them, the result is a wash of stage noise being fed into the front of house speaker system. This can lead to a really indistinct mix.

The technique of gating mics is designed to reduce the background noise level by setting a threshold which usually equates to roughly the smallest signal the mic is intended to pick up, and reducing the level of the channel for any signal lower than that threshold.



The Range is the amount the output has been reduced by, and this is user adjustable as well as the threshold.

Clearly, getting the right settings for both the threshold and the range are the main factors in reducing the amount of unwanted noise, but maintaining the integrity of the desirable sounds coming onto the mic.

If the threshold is set too low, there will be a very small difference in the noise present in the output. If the threshold is set too high, then quiet parts of the intended sounds from the mic will be vastly reduced in level, resulting in a "chopped" sound.

Like the compressors shown above, the other key parameters for the gate are its attack and release settings. The ATTACK time is usually in a range of about 0.0 ms–800.0 ms. This is the time from when the input signal exceeds the threshold level until the gate opens completely. The RELEASE time is usually in a range of about 0 ms–several seconds. It is the time over which the gate reaches its maximum effect.

Gates are very commonly used on Drums with great effect. This is due to the fact that Kicks, snares and toms in particular have transient type signals, which are much louder than the background, and have a fair amount of time (relatively speaking) between hits. It is this space between the hits that will be made significantly quieter using gates.

Typically the attack and release times for drums are kept very short (0.1 – 30 ms respectively), in order for the transient peak at the start of the hit to open the gate quickly, and allow the strength of the hit through.

As a starting point for those learning to use gates in live digital systems, I suggest recalling some of the presets available for the type of instrument or voice you are working with in each channel. Also experiment with the Threshold, Range, Attack and Release using the presets as a starting point, and you will soon be very comfortable making slight adjustments to improve the effectiveness of the gate.

As a bit of an acid test about how much noise you have saved, try using a pair of Headphones on a front of house mix with the gates set, and then temporarily disable or bypass the gates to hear the difference. You will wonder how you ever did without them!