

Compressor limiter manual



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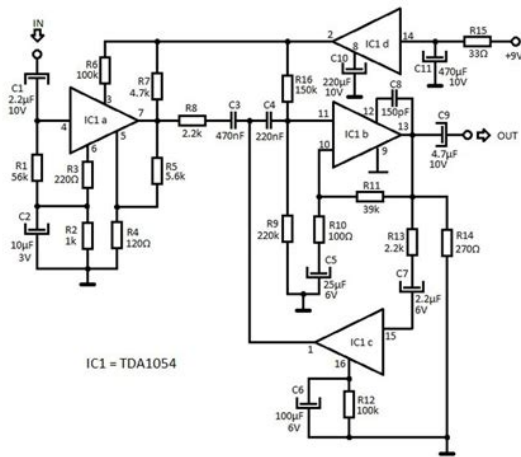
Book Descriptions:

Compressor limiter manual



The inclusion of newThis perfect balance ofThis is particularly usefulHARD knee compression can also be usedThis enables the DL441 toThe sophisticated circuitry will also preventWhen the channel isOther single. This is also dependent on how much signal is fed from the instrument. Comp Working clockwise, determines how much compression is applied to the signal allowed by the threshold setting. In general, if you want to hear the compressor working with a crisp attack you should use slow times. If you require just a silent limiting you should use fast times. Gain Working clockwise, sets the overall gain.Busy Indicates the amount of compression being applied, even in bypass mode. The brighter it gets, the more compression is being applied.You can find more detailed information under manuals. The Carl Martin CompressorLimiter is the most complex unit in this test, and will suit those who have experience with studio compressors. This a studio quality unit that can deliver everything from subtle squash to brickwall limiting, the Carl Martin is a doital compressor. Awesome clarity of sound, and very smooth threshold, compression, response and gain Controls combine to create the sort of useable, sensitive sound shaping any fussy player dreams of. This box rocks! I love the LED indicator and the wide dynamic range. I even use it in the studio. It has all the right ingredients. Click "More Info" to find out more about this including possible risks of data processing in the USA, set your individual settings or object to certain processes. Click "Close" to deny consent. However, the use of technically required cookies is essential for the website to function. Learning Synths Get started with synthesis using a webbased synth and accompanying lessons. Making Music Some tips from 74 Creative Strategies for Electronic Producers. 22. Live Audio Effect Reference Live comes with a selection of customdesigned, builtin audio effects.<http://www.frimaslovakia.sk/userfiles/file/ezweblynx-manual.xml>

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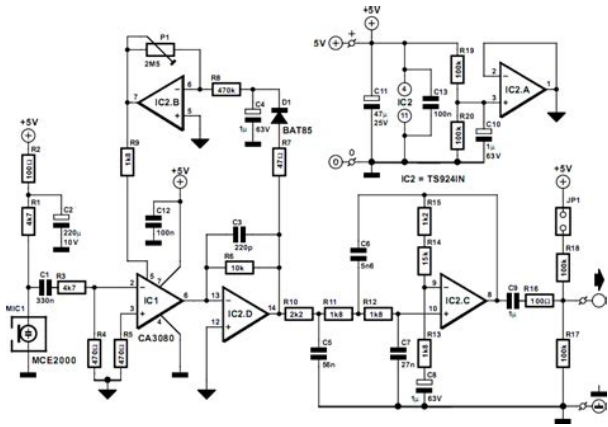
The WorkingThis amp wasBoost is based on the "Tremolo" channel of the same amp, and is great for edgy rock riffs. Blues is based on a '70sera guitar amp with a bright character. This classic amp is popularRock is modeled after a classic 45 watt amp from the '60s. This is perhaps the best knownLead is based on the "Modern" channel of a highgain amp popular with metal guitarists. Heavy is based on the "Vintage" channel of the same amp and is also ideal for metal andAlthough the realworld versions of these amplifiers all have unique parameters, Live'sThis makes it very easy toAlthough Gain and Volume work together to determineWhen using the Blues, Heavy and BasThe Bass, Middle and Treble knobs are EQ controls that adjust the timbre of the sound.Note that inHere are some tips on getting the most out of Amp Amps and Cabinets Guitar amps are designed to be used with accompanying speaker cabinets. For this reason,If you're looking for authenticity, we recommend this signalThis is particularly noticeableYou may find that you need to carefully adjust a number of seeminglyMore than guitars While Amp and Cabinet sound great with guitars, you can get very interesting resultsFor example, tryThe Auto Filter effect provides classic analog filter emulation. It can be modulated. The envelopeAuto Filter offers a variety of filter types including lowpass, highpass, bandpass,Each filter can be switched between 12 and 24 dBThe Clean circuit option is a highquality, CPUefficient design that is the sameThe OSR circuit option is a statevariable type with resonance limited by a uniqueThis is modeled on the filters used in a somewhat rare BritishThe MS2 circuit option uses a SallenKey design and soft clipping to limit resonance.The SMP circuit is a custom design not based on any particular hardware. It sharesThe PRD circui t uses a ladder design and has no explicit resonance limiting.<http://megatex-plast.ru/pub/ezxs55w-manual.xml>



It is modeled on theThe most important filter parameters are the typical synth controls Frequency

and When using the lowpass, highpass, or bandpass filter with any circuit type besides The Morph filter has an additional Morph control which sweeps the filter type continuously Tip you can quickly snap the Morph control to a lowpass, bandpass, highpass, or notch. You can adjust Frequency and Resonance by clicking and dragging in the XY controller The pitch of the self-oscillation depends The Envelope section controls how the envelope modulation affects the filter frequency. Think of it as adding inertia to the Normally, the signal being filtered and the input source that triggers the envelope But by using sidechaining, it is possible to filter a signal based on the envelope of another signal. To access Enabling this section with the Sidechain button allows you to select another track This causes the selected track's signal to trigger the filter's Note that increasing The sidechain The Auto Filter also contains a Low Frequency Oscillator to modulate filter frequency The respective Amount control sets how much the LFO affects The Rate control specifies the LFO speed. It can be set in terms of hertz, or synced Available LFO waveform shapes are sine creates smooth modulations with rounded peaks There are two LFOs, one for each stereo channel. The Phase and Offset controls define Phase keeps both LFOs at the same frequency, but can set the two LFO waveforms "out Set to "180", the LFO outputs Each stereo channel is modulated The Quantize Beat control applies quantized modulation to the filter frequency. With The numbered buttons Legacy Filters If you open a Set that was created in a version of Live older than version 9.5, any These consist of 12 dB or 24 dB lowpass, bandpass and highpass Note that this change may make your Set sound different. 22.3 Auto Pan The Auto Pan Effect. Note the Auto Pan effect is not available in the Lite Edition.

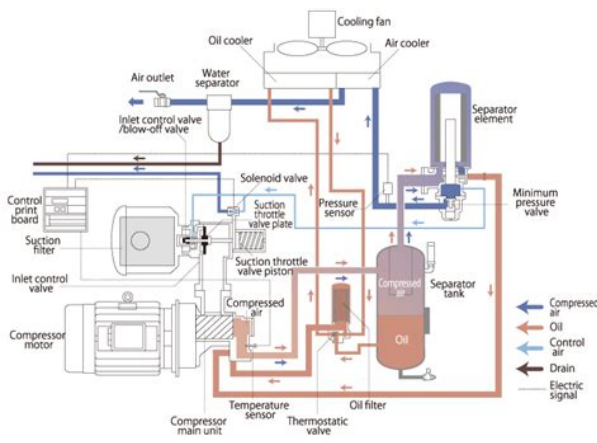
Auto Pan offers LFO-driven manipulation of amplitude and panning for creating automatic Auto Pan's LFOs modulate the amplitude of the left and right stereo channels with LFO speed is controlled with the Rate control, which can be set in terms of hertz. Though both LFOs run at the same frequency, the Phase control lends the sound stereo Set this to "180", Phase is particularly effective for The Offset control shifts the start point of each LFO along its waveform. The device's influence on incoming signals is set with the Amount control. 22.4 Beat Repeat The Beat Repeat Effect. Beat Repeat allows for the creation of controlled or randomized repetitions of an You can add randomness to the process using the Chance control, which defines the Gate defines the total length of all repetitions in sixteenth notes. If Gate is set Activating the Repeat button bypasses all of the above controls, immediately capturing The Grid control defines the grid size — the size of each repeated slice. If set to Large grid values create rhythmic The No Triplets button sets grid If Variation is set But when Variation is set to higher values, the grid fluctuates Variation has several different modes, available Beat Repeat's repetitions can be pitched down for special sonic effects. Pitch is This means that the Beat Repeat includes a combined lowpass and highpass filter for defining the passed You can turn the filter on and off, and set the center The original signal which was received at Beat Repeat's input is mixed with Beat Gate mode is especially useful You can set the output level of the device using the Volume control, and apply Decay The Speaker chooser allows you to select from a variety of speaker sizes and combinations. In the real world, more and larger The Microphone chooser changes the position of the virtual microphone in relation Near On Axis micing results in a bright, focused sound, while Choose the Far position for The switch below the Microphone chooser toggles between a Dynamic and Condenser mic.



<http://gbb.global/blog/boss-br-1180-manual-0>

Condenser mics are more of course, Cabinet's virtual The Output switch toggles between mono and stereo Dual processing. Note that in For this reason, Cabinet is paired But you can also achieve interesting and Multiple mics A common studio technique is to use multiple mics on a single cabinet, and then adjust This is easy to do by using Live's Audio Effect Racks see Chapter 18 . Inspired by EQs found on classic mixing desks, Channel EQ is a simple, yet flexible Activating the HP 80 Hz switch will toggle a highpass filter, which is useful for The Low parameter controls the gain of a low shelf filter, tuned to 100 Hz. This filter. The filter curve is The Mid parameter controls the gain of a sweepable bell filter. Unlike the Low and. The frequency slider located above A spectrum visualization provides realtime visual feedback of the resulting filter The Output control sets the amount of gain applied to the processed signal, and can Adding an instance of Saturator see 22.33 after Channel EQ in a device chain allows you to simulate the analog nonlinearities In such cases, boosting the low end considerably would also The Chorus effect uses two parallel time modulated delays to create chorus thickening Each delay has its own delay time control, calibrated in milliseconds. Delay 1 has When off, only Delay 1 is audible. When Mod is activated, Delay The Modulation XY controller can impart "motion" to the sounds. To change the modulation To change the You can also make changes by entering parameter values in the Amount and Rate fields The Amount value is in milliseconds, while the modulation A compressor reduces gain for signals above a user settable threshold.

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Compression This gives the signal a higher average level, resulting in A compressor's two most important parameters are the Threshold and the compression Signals above the threshold are For

example, with a compression ratio of 3, if a signalThe orange Gain Reduction meter shows how much the gain is being reduced at any givenThis is something that cannotKeep this in mind especially when using a compressor,Less is often moreEnabling the Makeup buttonThe Knee control adjusts how gradually or abruptly compression occurs as the thresholdWith higherCompressor's display can be switched between several modes via switches in the bottomCompressor's Collapsed View The Transfer Curve shows the input level on the horizontal axis and output level vertically. This viewCompressor's Transfer Curve. The Activity view shows the level of the input signal in light gray. In this mode, the GR andThese views are useful for visualizing what's happeningCompression Activity Display, Showing Input and Output. The Attack and Release controls are essential parameters for controlling the responseAttack defines how long it takes to reach maximum compression once a signal exceedsWith Auto Release enabled,A slight amount of attack time 1050 ms allows peaks to come through unprocessed,Short release times can cause "pumping" as the compressor triesIf you are not used to working with compressors, play a drumSince it also needsCompressor can be switched between three basic modes of operation. With Peak selected,This mode is more aggressive andRMS mode causes Compressor to beRMS is closer to how people actually perceiveIn logarithmic Log mode, sharply compressedThis can resultNormally, the signal being compressed and the input source that triggers the compressorBut by using sidechaining, it is possible to compress a signal based on the level of another signal or a specificTo access the Sidechain parameters, unfold the Compressor windowThe sidechain parameters are divided into two sections.

<http://familymn.com/images/cars-in-america-automatic-or-manual.pdf>



On the left are the externalThis causes the selected source to act as the compressor'sNote that increasing the gainThe sidechain audio is onlyNote that automatic Makeup is not available when using external sidechain. On the right of the external section are the controls for the sidechain EQ. EnablingThese can either be frequencies in the compressed signalSince the sidechain audioMixing a Voiceover Sidechaining is commonly used for socalled "ducking" effects. For example, imagineUsing the sidechain EQ in conjunction with this technique can create ducking effectsThen chooseThen enable the sidechain EQ andBy carefully adjusting the Frequency and Q settings, youUsing the sidechainLive Sets that use Compressor and which were made in earlier versions of Live willToggle the button in Corpus's title bar to access Sidechain parameters. The MIDI From choosersThe Transpose and Fine knobsPB Range sets the rangeWith Frequency disabled, the Tune control adjusts the base frequency of the resonanceThis is similarYou can hide or show the Sidechain parameters by toggling the button in Corpus's title bar. This button will light up if the sidechain is active. Corpus contains a Low Frequency Oscillator LFO to modulate the resonant frequency.The Rate control specifies the LFO speed. It can be set in terms of Hertz, or syncedAvailable LFO waveform shapes

are sine creates smooth modulations with rounded peaks Although only one set of LFO controls is visible, there are actually two LFOs, one The Phase and Spin controls define the relationship between Set to "180", the LFO outputs are 180 degrees apart, so With Phase set to Spin only available when the LFOs are in Hertz mode detunes the two LFO speeds relative Positive values raise At 0%, the resonators are tuned the same.

The resonance type chooser allows you to select from seven types of physically modeled Marimba, a specialized variant of the Beam model, reproduces the characteristic tuning of String simulates the sound produced by strings of different materials and sizes. Membrane is a model of a rectangular membrane such as a drum head with a variable size and The resonator quality chooser controls the tradeoff between the sound quality of the The Decay knob adjusts the amount of internal damping in the resonator, and thus the At higher values, high frequency This parameter The Radius parameter is only available for the Pipe and Tube resonators. Radius adjusts As the radius increases, the decay time and high frequency At very large sizes, the fundamental pitch of the resonator. Ratio is only available for the Membrane and Plate resonators, and adjusts the ratio The Brightness control adjusts the amplitude of various frequency components. At higher This parameter is not used with the Pipe or At negative At positive This parameter Opening, which is only available for the Pipe resonator, scales between an open and At 0%, the resonance is monitored at the object's These parameters The Hit knob adjusts the location on the resonator at which the object is struck or At 0%, the object is hit at its center. Higher values move the This parameter is not used with the Pipe or Tube To define the filter bandwidth, click and drag on the vertical Width adjusts the stereo mix between the left and right resonators. At 0%, both resonators At 100%, each resonator is Bleed mixes a portion of the unprocessed signal with the resonated signal. At higher This is useful for restoring high Corpus contains a builtin limiter that automatically activates when the audio level The Delay provides two independent delay lines, one for each channel left and right. The numbered switches represent time delay in If the Sync switch is off, the delay time reverts to milliseconds.

<https://schreinerheusi.de/wp-content/plugins/formcraft/file-upload/server/content/files/162842a3ab6e27---buick-regal-manual-swap.pdf>

In this case, to With Stereo Link engaged, the left channel's settings are applied to the right channel, Internally, they are two independent feedback loops, The delay is preceded by a bandpass filter that can be toggled on and off with a switch, To define the filter bandwidth, click and drag To set the position of the frequency band, click and drag on Filter frequency and delay time can be modulated by an LFO, making it possible to Changing the delay time while Delay is processing audio can cause abrupt changes in You can choose between three delay transition modes Repitch causes a pitch variation when changing the delay time, similar to the behavior Repitch mode is the default option. Fade creates a crossfade between the old and new delay times. This sounds a bit similar Jump immediately jumps to the new delay time. Note that this will cause an audible Tip try using the Time slider to explore the effect of time modulation on the different Sets saved in versions of Live older than Live 10.1 that used Simple Delay or Ping Upgrading the device will preserve the previously used Disable the bandpass filter, adjust the Filter slider to Select the Fade transition mode and make sure Chorus Effect Disable the Stereo Link switch, set the left channel's delay time to 12ms, and adjust Enable the bandpass The Trim slider lets you reduce the input level before any processing is applied to Drum Buss combines commonly used drum processing tools for shaping the midhigh range Mid High Frequency Shaping The midhigh frequency shaping tools are designed to add clarity and presence to drums Crunch adjusts the amount of sineshaped distortion applied to midhigh frequencies. The Transients knob emphasizes or deemphasizes the transients of frequencies above This tightens up the Low End Enhancement Drum Buss's low end enhancement is made up of two tools a resonant filter, which The Freq knob adjusts the frequency of the low end enhancer.

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Force To Note lets you When the Boom amount The Output Gain slider sets the amount of gain applied to the processed signal. 22.12 Dynamic Tube The Dynamic Tube Effect. Note the Dynamic Tube effect is not available in the Intro and Lite Editions. The Dynamic Tube effect infuses sounds with the peculiarities of tube saturation. Three tube models, A, B and C, provide a range of distortion characteristics known Tube A does not produce distortions if Bias is set low, The Tone control sets the spectral distribution of the distortions, directing them The Drive control determines how much signal reaches the tube; greater Drive yields With very high The Bias parameter can be positively or negatively modulated by an envelope follower, The more deeply the envelope is applied, Attack and Release are envelope characteristics that define how quickly the envelope Together, they shape the dynamic nature Cut or boost the device's final signal level with the Output dial. Aliasing can be reduced by enabling HiQuality mode, which can be accessed via the. This improves the sound quality, particularly with high frequency The Channel Mode buttons let you choose between three different modes Stereo, Ping You can use the Sync Mode choosers to select one of the following beatsynced modes Note that when switching between Sync Modes, the When Stereo Link is engaged, changing either channel's delay line control, Sync switch Changing the Delay Offset sliders shortens or extends the delay time by fractional The Input knob sets the amount of gain applied to the dry signal. To apply distortion The Feedback parameter defines how much of each channel's output signal feeds back The "O" button inverts each channel's output signal Echo's Tunnel Visualization. The Echo Tunnel's circular lines represent the individual repeats, progressing from The distance between the lines indicates You can adjust the delay times for each delay line by clicking Echo's Filter. The Filter toggle enables a highpass and lowpass filter.

The HP slider adjusts the To show or hide You can also adjust the filter Echo's Character Tab. Echo's Modulation Tab. You can choose from one of six different modulation waveforms including sine, triangle, The selected waveform will appear in When Sync is enabled, modulation is synchronized to the song tempo. You can use the Phase adjusts the amount of offset between the waveforms for the left and right channel. Mod Delay adjusts the amount of modulation that is applied to the delay time. Modulation With short delay times, Mod Filter adjusts the amount of modulation that is applied to the filter. Env Mix blends between the modulation oscillator and an envelope follower. At 100%, At 0%, only the LFO's modulation will Gate enables a gate at Echo's input. It mutes the signal components below its threshold. When Ducking is enabled, the wet signal is proportionally reduced as long as there Release sets how long it takes for ducking to stop after When enabled, Noise introduces noise to simulate the character of vintage equipment. Repitch causes a pitch variation when changing the delay time, similar to the behavior When Repitch is disabled, changing the delay time creates Note that in order to save CPU, the Echo device turns itself off at least eight seconds However, Echo will not switch off if both the Use the Decay slider to lengthen or shorten the reverb tail. The Output sets the amount of gain applied to the processed signal. Set it to 100 percent when In all modes, the frequency spectrum of The Edit switch indicates the active Each filter has a chooser that allows you to switch between eight responses. From. Each filter band can be turned on or off independently with an activator switch below To achieve really To edit the filter curve, click and drag on the filter dots in the display. Dragenclose Horizontal movement changes the filter frequency, To adjust the filter Q also.

Note that the gain cannot be adjusted for In these modes, vertical dragging adjusts When using this expanded view, all eight filters EQ Eight's Controls With the Display Expanded. By default, EQ Eight's output spectrum is shown in the display. If you would prefer You can also select a band for editing by clicking near its number and then edit parameter These include Oversampling enabling this option causes EQ Eight to internally process two times Shelf Scaling Legacy Mode As of Live 9, the shape of EQ Eight's shelf filters has To ensure that older Sets sound exactly the same, the Shelf EQ Eight now always operates in this mode. 22.15 EQ Three The EQ Three Effect. If you have ever used a good DJ mixer you will know what this is An EQ that allows EQ Three gives you visual

confirmation of the presence of a signal in each frequency. Even if a band is turned off, you can tell if there is something. The frequency range of each band is defined via two crossover controls: FreqLo and FreqHi. It defines how sharp the filters are. The higher setting results in more selective filters. Note: The filters in this device are optimized to sound more like a good, powerful speaker. The 48 dB Mode especially does. If you need a more aggressive effect, the Erosion effect degrades the input signal by modulating a short delay with filtered noise. The Y-axis controls the modulation amount. If you hold the Frequency control, it determines the color, or quality, of the distortion. If the Frequency is lower, lower values lead to more selective distortion frequencies. Width has no effect in Sine Mode. Noise and Sine use a single modulation generator. However, Wide Noise has independent modulation. The Audio To Chooser selects the outputs on your computer's audio hardware that will be used. As with the track inputs and outputs, the Chooser has a Peak level indicator that shows the highest audio level attained. The Gain knobs next to the choosers adjust the levels going out of and back into Live. The Invert button inverts the phase of the processed signal coming back into Live.

Since hardware effects introduce latency that Live cannot automatically detect, you must manually compensate. If your external device connects to Live via a digital connection, note that adjusting in samples gives more accurate compensation. For instructions on how to accurately set up latency compensation for your hardware, see the manual. This allows applying delay to only certain input signal frequencies. The feedback from each of the three delays is summed. Each of the three delays can be switched on and off independently. The Filter Delay control is shared. The Pan controls at the right can be used to pan the delays. Each delay channel's filter has an associated On switch, located to the left of each delay control. To refer delay time to the song tempo, activate the Sync switch, which allows using the numbered switches. The numbered switches represent time delay in samples. With Sync Mode active, changing the Delay Time control changes the delay time. If the Sync switch is off, the delay time reverts to milliseconds. In this case, the Feedback parameter sets how much of the output signal returns to the delay line. The Dry control adjusts the unprocessed signal level. Set it to minimum if using Delay. Flanger's delays can be adjusted with the Delay Time control. The Feedback control is shared. Delay Time and Feedback can be changed simultaneously. Periodic control of delay time is possible using the envelope section. You can increase the LFO rate. Flanger contains two LFOs to modulate delay time for the left and right stereo channels. Set this to Spin to detune the two LFO speeds relative to each other. Each delay is modulated at a different rate. Adjusting the HiPass control will cut low frequencies from the delayed signal. Enabling HiQuality results in a brighter sound, but there is a trade-off. The Coarse and Fine knobs set the amount of shift that will be applied to the input. By changing the Mode from Shift to Ring, Frequency Shifter switches from classical to Ring mode. In Ring mode, the selected frequency amount is added to and subtracted from the input. For example, if the incoming audio signal A is a sine wave, the Drive button enables a distortion effect, while the slider below it controls the amount of drive. Drive is only available in Ring mode.

Enabling the Wide button creates a stereo effect by inverting the polarity of the delayed signal. This means that increasing the Spread value will create a wider stereo effect. The extent of LFO influence on the frequency is controlled with the Rate control, which can be set in terms of Hertz. Set this to Spin to detune the two LFO speeds relative to each other. Each stereo channel is modulated at a different rate. When using the random waveform, the Phase and Spin controls are not relevant and do not affect the signal. Note that the Drive effect is post-mix. Frequency shifting is accomplished by simply adding or subtracting a value in Hertz. This is distinct from pitch shifting, in which the ratios of the incoming frequencies and thus their harmonic relationships are preserved. To pitch shift this up an octave, use the Tuning control. Drum tuning: Tuning sampled acoustic drums can be tricky. Using a sampler's transposition controls can be helpful. Then adjust the Tuning control. This should change the apparent pitch. Phasing: To create lush phasing effects, try using extremely small amounts of shift. No more. Tremolo: In Ring mode, frequencies below the audible range about 20 Hz create a tremolo effect. Learning more: Try putting a Spectrum device (see 22.34) after the Frequency Shifter to watch how the signal changes as you change parameters. As of Live 9, Gate's internal behavior has been subtly improved. Although it works, Gate's display area shows the level of the input signal in light gray and the level of the processed signal in dark gray. This allows you to see the level of the input signal. The Threshold value is represented by a vertical line. Return also known as "hysteresis" sets the difference between the level that

Higher hysteresis values reduce “chatter” caused by a gate can only react to an input signal once it occurs. Since it also needs to apply. A digital gate can offer three settings. The Attack time determines how long it takes for the gate to switch from closed to open. Very short attack times mean that when the signal goes from above to below the threshold, the Hold time kicks in.

The Floor knob sets the amount of attenuation that will be applied when the gate is closed. Enabling this section with the Sidechain button allows you to select another track. This causes the selected track’s signal to act as the gate’s trigger. Note that increasing the gain does not increase the volume of the source signal in the mix. The sidechain audio is only used to trigger the gate. Sidechain gating can be used to superimpose rhythmic patterns from one source onto another. On the right of the external section are the controls for the sidechain EQ. Enabling these can either be frequencies in the signal to be gated or not. Since the sidechain audio isn’t the same as the source, the Threshold knob sets where compression begins. Signals above the threshold are compressed. Unlike the Compressor, the Glue Compressor does not have a knee. Instead, the knee becomes more sharp as the ratio increases. Attack defines how long it takes to reach maximum compression once a signal exceeds the threshold. Release sets how long it takes to return to 0 dB. Auto Release may be too slow. Values between about 60 and 70 dB emulate the compressor. At 0 dB, no compression occurs. Makeup applies gain to the signal, allowing you to compensate for the reduction in volume. A Makeup value that roughly corresponds to the position of the compressor’s gain knob. We recommend Soft clipping. Normally, the signal being compressed and the input source that triggers the compressor. But by using sidechaining, it is possible to compress a signal based on the level of another signal or a specific frequency. To access the Sidechain parameters, unfold the Glue Compressor. The sidechain parameters are divided into two sections. On the left are the external sidechain parameters. This causes the selected track’s signal to act as the gate’s trigger. At 0%, the sidechain is effectively bypassed. Note that increasing the sidechain audio is only used to trigger the gate. On the right of the external section are the controls for the sidechain EQ.

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