Ableton plugins manual

Windows and MacReference Guide by Dennis DeSantis, Michelle Hughes, Ian Gallagher, Kevin Haywood, Rose Knudsen, Gerhard Beehles, Jacob Range, Robert Henke, Torsten Salama. Schonhauser Alli 6-7 | 10119 Berlin, GermanyNumber Another: www.ableton.com/support/contact Content provided by: SONiVOX - www.sonivoxrocks.comChocolateChocolate Audio - www.chocolateaudio.comPuremagnetik — www.puremagnetik.comCycling '74 — www.cycling74.comSonArte — www.sonarte.cae-instruments — www.e-instruments.comZero-G — www.e-instruments.comZero-G — www.cycling74.comSonArte — www.sonarte.cae-instruments — www.e-instruments.comZero-G — www.e-instruments.comZero-G — www.e-instruments.comZero-G zero-g.co.ukGoldbab - www.goldbaby.co.nzSample Magic — www.samplemagic.comSoniccouture — www.soniccouture.comLoopmasters — www.loopmasters.comUppercussion — www.uppercussion.com www.applied-acoustics.com. Working with tools and effects chapter (see chapter 17) explains the basics of using effects in Live.22.1 ampM. (Note: Amp effect is not available in the introduced versions, latin, and metrics.) Amp is an effect that mimics the sound and character of seven classic guitar speakers. Developed in collaboration with Softube, Amp uses physical modeling technology to provide a range of original and usable speaker tones, with a simple and consistent set of controls. There are seven amp models to choose from: clean is based on a brilliant channel The classic amp of the '60s. This amp was widely used by guitarists from British Invasion.Boost based on the Tremolo channel of the same amp, and great for edgy rock riffs. The blues is based on the '70s era of amp guitar with a bright character. This classic amp is popular with country, rock and blues guitarists. Rock-style 45-watt classic of the '60s. This is perhaps the best-known rock amp of all time. Lead is based on a modern channel of high-gain amp popularity with metal guitarists. Heavy is based on a vintage channel of the same amp, which is also ideal for metal and gong sounds. Bass-style after pa is rare from the '70s which has become popular with bass players due to its strong low end and fluff in high sizes. Although the real-world versions of these speakers all have unique parameters, Amp Live's effect uses the same set of controls for each model. This makes it very easy to change the overall character of the sound quickly without the need for numerous modifications. Gain adjusts the input level to the preamplifier, while the volume adjusts the output stage of the power amplifier. Although profit and size work together to determine the total level of Amp, Gain is the primary control of the distortion amount. Higher gain settings cause more distorted sound. When using blues, heavy and bass models, high sound levels can also add great distortion. Bass knobs, middle and triple are EQ controls that set the sound bell. As in a real-world amplifier, amp EQ parameters interact with each other — and with other amp parameters — in sometimes unlinear and sometimes unexpected ways. For example, eq levels can increase, in some cases, also increase the amount of distortion. Presence is an additional medium/high-frequency sound control in the power amp phase. Its effect on sound varies greatly depending on the user's amp model but can add (or subtract) an edge or crisp. Switch output between mono and stereo (dual) processing. Note that in dual mode, Amp uses the CPU twice. Dry/wet control adjusts the balance between treated and dry signals.22.1.1 amp-style amp supperations on analog devices in the real world, their behavior may sometimes be difficult to predict. Here are some tips on getting the most out of Amp: Amp and Guitar Cabinets are designed to be used with accompanying speaker cabinets. For this reason, Amp comes with a companion effect called Cabinet (see 22.5) which is designed to be used after amp in a hardware series. If you are looking for authenticity, we recommend this signal flow. But you can also achieve interesting and strange sounds using amp and cabinet independently. Different electrical circuits in guitar amps work with a constant and steady amount of electricity. For this reason, even a particular parameter may inadvertently reduce the amount of energy available elsewhere in the amp. This is special. In EQ controls. For example, running the trilogy can reduce the level of bass and medium-range frequencies. You may find that you need to carefully adjust the number of parameters that seem irrelevant to get the results you want. More than guitarsWhile amp and great sound cabinet with guitars, you can get very interesting results by feeding them with drums, synthesizers or other sound sources. For example, try using amp with the operator (see 24.6) or analog (see 24.1) to add representative grit to digital sounds.22.2 Automatic filtering provides automatic filtering effect. Embed by envelope follower and/or financial tire system can be edited to create animated filter effects. The envelope follower can track either the filtered signal or an external side source. Auto Filter offers a variety of filter types including low lane, highpass, pass, grade, and morph filter. Each filter can be switched between 12 and 24 decibel slopes as well as a selection of analog circuit-style behaviors developed in conjunction with Cytomic that mimic hardware filters found on some classic analog synthesizers. The clean circuit option is a high-quality design, the CPU is effective and is the same filters used in EQ Eight (see 22.14). This is available for all types of filters. The OSR circuit option is a variable state type with a limited echo by a uniquely hard-to-cut diode. This is similar to the filters used in a fairly rare British monosynth and is available for all types of filtering. The MS2 circuit option uses sallen-Key design and soft clip to reduce resonance. It is similar to filters used in the famous Japanese mono semi-modular and available for lowpass and highpass filters. The SMP circuit is a custom design that is not based on any particular device. It shares the properties of both MS2 and PRD circuits and is available for lowpass and highpass filters. The PRD circuit uses a ladder design and has no explicit echo limit. It is similar to the filters used in a dual oscillator monosynth from the United States, and is available for lowpass and highpass filters. The most important filter parameters are typical synth controls frequency and resonance. The frequency determines where the filter is applied in the harmonic spectrum; When using a lowpass, height, or filter band bar with any circuit type besides clean, there is an additional engine control that can be used to add gains or distortion to the signal before it enters the filter. Morph filter has an additional control of the mutants that continuously sweep the filter type from the low lane to the range lane to the high lane to the slit and back to the lowpass. Tip: You can quickly snap-control morph to the low corridor, bar, height, or set up the slit via custom options right in click on (PC) / CTRL-click (Mac) context list of the shape handle. You can adjust the frequency and resonance by clicking and dragging in Control or by knobs. You can also click on Frick and Digital Res screens and type exact values. When using non-clean circuit types, self-oscillation resonance control sits. In the ringing values above 100%, the filter will continue to ring indefinitely even after the input signal is turned off. Self-oscillation depends on both frequency and resonance values. The envelope section controls how the envelope is modified affects the filter frequency. The Amount control determines how the envelope affects the filter frequency, while the offensive control sets out how the envelope responds to increased input signals. Low attack values cause a rapid response to input levels; Think of it as adding rigidity to the response. Low version values cause the envelope to respond more quickly to falling input signals. The higher values expand the envelope decay. Typically, the filtered signal and the input source that triggers the envelope child are the same. But using sidechaining, it is possible to filter the signal based on the envelope from another signal. To access sidechain parameters, an automatic filter window unfolds by switching the button in the address bar. Enabling this section with the Sidechain button allows you to select another path from the selectors below. This causes the specified route signal to run the filter envelope follower instead of the signal that is actually filtered. The gain handle adjusts the level of external sidechain input, while the dry/wet handle allows you to use a combination of sidechain and original signal as a dependent envelope operator. With dry/wet at 100%, follow the sidechain source envelope follower exclusively. At 0%, the side chain is effectively exceeded. Note that increased profit does not increase the size of the source signal in the mix. The sidechain audio is only a trigger for the envelope follower and has not actually been heard. An automatic filter also has a low-frequency oscillator to periodically modify the filter frequency. The Amount control determines how much LFO affects the filter. This can be used in conjunction with or instead of a dependent envelope. The control determines the LFO speed rate. It can be set in terms of Hertz, or synchronized with the frequency of a song, allowing control of rhythmic filtering. The waveforms available for FO are pocket (creates smooth formations with rounded peaks and valleys), square, triangle, sawtooth, sawtooth down, sample and holding (generates positive randomness and negative adjustment values) in mono stereo. There are two LFOs, one for each stereo channel. Phase and Offset controls define the relationship between these two LFOs.Phase keeps both than less compressed material. This can result in smoother and less noticeable compression of Lin mode. Note that the Lin/Log switch is not visible in the folded compressor view. A compressor with Sidechain Section.Normally, the signal is compressed and the input source that releases the compressor is the same signal. But using the sideline, it is possible to compress a signal based on another signal level or a specific frequency element. To access sidechain parameters, the compressor window unfolds by switching the button in the address bar. The side chain parameters are divided into two parts. On the left are external controls. Enabling this section with the Sidechain button allows you to select any of the live internal routing points (see 14.6.1) from the selectors below. This leads to the specified source as a compressor operator, rather than the signal that is already compressed. The gain handle adjusts the level of external sidechain input, while the dry/wet handle allows you to use a combination of sidechain and original signal as the compressor operator. With dry/wet at 100%, the compressor is completely operated by a sidechain source. At 0%, the side chain is effectively exceeded. Note that increased profit does not increase the size of the source signal in the mix. The sidechain sound is just a compressor player and has not actually been heard. Note that automatic makeup is not available when using an external sidechain. To the right of the outer section are the controls of the sidechain EQ. Enabling this section leads to the compressor being operated by a certain range of frequencies, rather than a complete signal. These can be either frequencies in the compressed signal or, using EQ in conjunction with an external side chain, frequencies in another path The headphone button between the external sections and EQ allows you to listen to only sidechain inputs, bypassing the compressor output. Since sidechain sound is not fed to the output, and is only the trigger for the compressor, this temporary listening option can make it much easier to set sidechain parameters and hear what actually makes compressor action.22.8.2 Pressure TibbsThis section offers some tips for using the compressor effectively, especially with sidechain options. Mixing VoiceoverSidechaining is commonly used for so-called dodging effects. For example, imagine that you have one track that has a sound and another track that contains background music. Since you want voice over to be the top source in the mix, background music must get out of the way every time the narrator speaks. To do this automatically, insert Compressor to the music track, but select output of the narration path as an external source. Sidechaining/ducking is a secret weapon dance music producer because it can help ensure that basslines (or even whole mixes) always make room for kick drum. By inserting a compressor on the bass track (or master) and using a kick drum track as a sidechain input, you can help control problematic low frequencies that may interfere with the kick drum attack. Using sidechain EQ in conjunction with this technique can create dodging effects even if you only have a mixed drum track to work with (rather than an isolated kick drum). In this case, enter the compressor on the path you want to duck. Then choose the drum track as an external sidechain source. Then enable EQ from EQ and choose a lowpass filter. By carefully adjusting the frequency and Q settings, you should be able to isolate the kick drum from the rest of the roller mix. Using sidechain listening mode can help you adjust EQ until you find the settings you're happy with.22.8.3 The upgrade of Legacy ModeCompressor's internal algorithms has been updated in Live 9, in collaboration with Dr. Joshua D. Reese of the Digital Music Center, and Queen Mary University of London.Live Group that uses a compressor that has been made in previous versions of Live will show the upgrade button in the title bar of each instance of the compressor set in Live 9. Press the upgrade button in order to upgrade this compressor example to the latest, and improve algorithms. Note that this may lead to your set to a different sound.22.9 Corbusedge effect. (Note: The Corpus effect is not available in the introduced, standard and light versions.) The Corpus is an effect that mimics the acoustic properties of seven types of resonant objects. Developed in collaboration with applied acoustic systems, Corpus uses physical modeling technology to provide a wide range of parameters and modulation options. The frequency and/or decay rate of midi resonance can be embedded, by enabling frequency and/or off decay In the side chain section. Switch the button in the address bar in Corpus to access sidechain parameters. MIDI of selectors allow you to select the MIDI path and touch the point through which to receive MIDI note information. With frequency enabled, the resonance adjustment is determined by the incoming MIDI note. If multiple notes are held at the same time, the last/low switch determines whether another or lower note will be a priority. The switch and fine knobs allow for a snout and a fine compensation of midi-modulation tuning. The PB band determines the range in half of the adjustment tone of the pitch bend. With frequency disabled, Tune control adjusts hertz's primary resonance frequency. The corresponding MIDI note number and exact adjustment compensation are displayed below. Enabling stop decay causes MIDI to note off messages to mute the echo. The slider down the switch determines how far the MIDI note is off the echo mute messages. At 0%, a note is ignored and the decomposition time is based solely on the value of the join parameter that falls under the ring type limit. This is similar to how hammer instruments in the real world such as marimba and glockenspiels behave. At 100%, the resonance is immediately mute in the off note, regardless of the decay time. You can hide or show Sidechain parameters by switching the button in the Corpus address bar. This button will light up if the side string is active. Corpus has a low frequency oscillator (LFO) to modify the resonant frequency. The Amount control determines how much the system specified for LFO affects the frequency. The control determines the LFO speed rate. It can be set in terms of Hertz, or synchronized with the rhythm of the song, allowing for controlled rhythmic formation. The waveforms available for FO are pocket (creates smooth formations with rounded peaks and valleys), square, triangle, sawtooth, sawtooth down and two types of noise (stepped and smooth). Although only one set of LFO controls is visible, there are already two LFOs, one for each stereo channel. Phase and spin controls determine the relationship between these two LFOs.Phase (available only when LFOs is synchronized for the frequency of the song) both LFOs retains in the same frequency, but lfo wavelengths can be set out of the stage with each other, creating stereo motion. Set to 180, LFO outputs are 180 degrees apart, so that when one LFO reaches its peak, the other is minimal. With a set phase to 360 or 0, two LFOs run in sync. Spin (only when LFOs is in Hertz mode) is coupled with two LFO speeds relative to each other. Each stereo channel is adjusted at a different frequency, as determined by the amount of spin. For longitudinal noise, phase and spin controls are irrelevant and do not affect sound. DeTunes posted two resonant regarding each other. Positive values raise the pitch from the left resonant while the pitch is reduced from the right one, while the negative values do the opposite. At 0%, resonant is tuned The chosen type resonance allows you to choose from among seven types of physically-stylere objects: a beam simulates the resonance properties of beams of different materials and sizes. Marimba, a specialized variant of the beam model, reproduces the characteristic tuning of marimba bar tones that are produced as a result of cutting a deep arc of bars. Simulates the sound chain produced by strings of different materials and sizes. The membrane is a model of a rectangular membrane (such as a drum head) with variable size and construction. The panel simulates the production of sound by a rectangular panel (flat surface) of various materials and sizes. The pipe simulates a fully open cylindrical tube at one end and has a variable opening at the other end (modified with the opening parameter.) a tube simulates a fully open cylindrical tube at both ends. The resonant quality selector controls the trade-off between the sound quality of the resonant and the performance by reducing the number of tones that are calculated. Basic uses minimal CPU resources, while the entire creates a more sophisticated echo. This parameter is not used with pipe or tube resonants. The decay handle adjusts the amount of internal damping in the resonant, thus decay time. The material handle adjusts the difference of damping at different frequencies. When values are low, low-frequency components are degraded slower than high frequency components (which mimic objects made of wood, nylon or rubber). At the top of values, high-frequency components decompose more slowly (which mimics objects made of glass or metal). This parameter is not used with pipe or tube resonants. Radius parameter is only available for pipe and pipe ranouts. The radius of the tube or tube is adjusted. As the radius increases, the decay time and the high frequency keep both increasing. In very large sizes, the basic pitch of resonant changes too. Decay parameters and materials/radius can also be controlled with the X-Y controller. The ratio is only available for the membrane and resonant panel, and adjusts the size ratio of the object along its X-Y axes. Brightness control adjusts the capacity of different frequency components. At the highest values, the higher the frequencies are higher. This parameter is not used with pipe or tube resonants. It's a sin. (Inharmonics) adjusts the pitch of resonant harmonics. In negative values, frequencies are compressed, increasing the amount of low particles. In positive values, frequencies are extended, increasing the amount of higher molecules. This parameter is not used with pipe or tube resonants. Open, which is only available for pipe nanons, scales between open and closed tube. At 0%, the pipe is completely closed on one side, while at 100% the pipe is open at both ends. The L and R thumbnail controls adjust the location on the right and left resonance device where vibrations are measured. At 0%, the echo is monitored in Center. The higher values move the listening point closer to the edge. These parameters are not used with pipe or tube resonants, which are always measured in the middle of their permanently open end. The Hit handle adjusts the location on the ringdevice where the object is struck or otherwise activated. At 0%, the object is multiplied at its center. The higher values move the activation point closer to the edge. This parameter is not used with pipe or tube resonants. The processed signal is fed by a lowpass and highpass filter that can be controlled with the X-Y controller. To define the filter bandwidth, click and drag on the vertical axis. To set the position of the frequency band, click and drag on the horizontal axis. The filter can be turned on or off with the filter key. The display adjusts the stereo mix between the left and the right. At 0%, both resones are fed evenly for each side, resulting in monoproduction. At 100% each resonant is sent exclusively to one channel. Liquefaction blends part of the unprocessed signal with a friendly signal. At the top of the values, more of the original signal is applied. This is useful for high frequency recovery, which can often be serviced when the adjustment or quality is set to low values. This parameter is not available with pipe or tube resonants. Gain enhances or reduces the level of the treated signal, while dry/wet control adjusts the balance between the dry input signal and the signal sent to the Corpus process. Turning dry/wet down will not cut the resonance that is currently sounding, but instead of stopping the new input signals from processing. Corpus has an embedded selector that activates automatically when the volume is too high. This is referred to by the LED in the upper right corner of the delayThe Delay Effect.The delay provides two independent delay lines, one for each channel (left and right). To indicate the delay time to the frequency of the song, activate the sync key, which allows the partition picker to be used in the delay time. Numbered switches represent time delay in the 16 notes. For example, selecting 4 delays the signal with four 16 notes, which is equal to one pulse (quarter of a note) of delay. If the sync key is turned off, the delay time returns to milliseconds. In this case, to free up the delay time, click and drag the delay time handle. With the sharing stereo link, the left channel settings are applied to the right channel, either changing the channel sync code or delaying the time settings to apply the changes on both sides. The note parameter determines how much each output signal feeds back into the delay line input. Internally, they are two independent feedback loops, so the signal on the left channel does not feed back to the right channel and vice versa. The button will delay the audio cycle that is in the buffer at the moment you press the button, and ignore any new entry until the freeze is disabled. The delay is preceded by Filter that can switch and stop with switch, and control it with the X-Y controller. To define the filter bandwidth, click and drag on the vertical axis. To set the position of the frequency band, click and drag on the horizontal axis. Filter frequency and delay time can be adjusted by the financial tire system, making it possible to achieve a range of sounds from light chorus-like effects to heavy noise. A modified slider determines the frequency of hertz's modulation oscillator. The filter slider adjusts the amount of the configuration that is applied to the filter, and the Time slider adjusts the amount of the configuration that is applied to the delay time. Changing the delay time while processing the sound delay can cause sudden changes in the late signal sound. You can choose between three delay transition modes:Repitch causes pitch variation when the delay time changes, similar to the behavior of the old tape delay units. Repitch mode is the default option. Fade creates an intersection between old and new delays. This looks a little similar to the stretch time if the delay time is gradually changed. Jump jumps instantly to the new delay time. Note that this will result in an audible click if the delay time changes during delays. Tip: Try using the time slider to explore the effect of time adjustment on different transition modes. When the Ping Pong key is activated, the signal moves from the right output to the left. Dry/wet control adjusts the balance between treated and dry signals. Set it to 100 percent when you use a delay in a return path. The dry/wet parameter context menu lets you switch equal sound. When enabled, the 50/50 mix sound will be equally loud for most signals. Saved sets in live-age versions of Live 10.1 that use simple delay devices or ping pong delays will show an upgrade button in the address bar for each instance of delay when downloading set. Upgrading the device will maintain the free delay time range of the previously used device, and will only affect set sound or advance if the free delay time parameter is set either to macro control or to the Max Live.22.10.1 TipsGlitch EffectEnable delay switch switch link and stereo connection time set to about 400-500ms. Ask for notes to 80% or higher. Disable the bandpass filter, adjust the filter slider to 0%, and set the time slider to 100%. Select the go-go mode and make sure that ping pong is off. Set dry/wet control to 80% or higher. EffectDisable chorus switch stereo link, set the left channel delay time to 12ms, and adjust the right channel delay time to 17ms. Enable the range pass filter, adjust the filter frequency to 750 Hz, and adjust the view slider to 6.5. Set the rate slider to 5 Hz, bring the filter slider to 10%, and request the time slider to 12%. Select Repitch Transmission Mode and Enable Ping Pong Switch.22.11 Drum Bus (Note: Bus Drum Effect Is Not Available Introduced and yi.) Drum Buss is an analog-style drum processor designed to add body and craft to a range of drums, while sticking them together in a tight mix. The Trim slider lets you reduce the input level before applying any processing to the signal. Comp Switch applies a fixed compressor to the input signal before it is processed by distortion. The compressor is optimized to balance the sets of drums, with fast attack, medium release and moderate adjust ratio, as well as ample makeup gains. There are three types of distortion that can be applied to the input signal. Each distortion type adds an increasing degree of distortion, while lending its own character to the public sound: Soft: Waves DistortionMedium: Reducing hard distortion: distortion of the pieces with bass reinforcement for more density, it is possible to push the input before distorting it. Drive lets you determine how much drive is applied to the input signal. Drum Boss combines drum processing tools commonly used to form a high medium band and fill the low end, which we'll look at in the following sections. Mid-high-frequency ShapingThe mid-frequency forming tools are designed to add clarity and presence to drums such as traps and hi hats. The pocket-shaped distortion-shaped quantity crisis adjusts applied to high medium frequencies. Damp control is a low pass filter, which removes unwanted high frequencies that can occur after adding distortion. A transient handle confirms or de-confirms transients of frequencies above 100 Hz. Positive values add attack and increase maintain, resulting in full, punchy sound. Negative values also add attack, but low preservation. This pulls up the drums, giving them a clearer, more crisp sound with less space and a scintillating. Low-endDrum Enhancement Buss low-end-end enhancements consists of two instruments: a resonant filter, which significantly enhances bass frequencies, as well as decay control, which allows you to adjust the decay rate for both incoming sound and signals processed by a resonant filter. These tools help you fill the low end of your drums. The boom handle adjusts the amount of low reinforcement produced by the resonant filter. The bass scale lets you see the effect of the boom on the signal, which can be particularly useful if you can't hear it. The team handle adjusts the improved low end frequency. Force To Note lets you adjust a low-end enhancer by setting its frequency to the value of the nearest MIDI note. Decay control adjusts the low frequency impairment rate. When the surge amount is set to 0%, the decay only affects the incoming signal (post-drive distortion and distortion). When the boom level is set above 0%, the decay affects both the incoming and processing signals. For an individual low frequency optimizer result, enable the test boom via the headset icon. OutputThe dry/wet control adjusts the balance between processing and dry The output gain slider adjusts the amount of gain applied to the processed signal.22.12 TubeThe Dynamic TubeThe Effect. (Note: Dynamic tube effect is not available in introduced and latin versions.) The dynamic tube effect transmits sounds with tube saturation properties. The integrated envelope follower creates dynamic tonal variations related to the input signal level. Three tube models, A, B and C, provide a range of known distortion properties of real speaker tubes. Tube A does not produce distortions if the bias is set low, but will kick in whenever the input signal exceeds a certain threshold, creating bright harmonics. Tube C is a very poor tube amp that produces distortions all the time. The qualities of Tube B lie somewhere between these two extremes. Adjust the tone set the spectral distribution of distortions, and direct them to higher records, or through the medium and deeper range. The drive control determines the amount of signal that reaches the tube; The density of the tube is controlled by a bias disc, which pushes the reference to the famous worlds of nonlinear distortion. With very high amounts of bias, the signal will really start to break. The Bias parameter can be positively or negatively embedded by an envelope follower, which is controlled with the envelope handle. The deeper the envelope is applied, the higher the bias point is affected by the input signal grade. Negative envelope values create expansion effects by reducing distortion on loud signals, while positive values will make loud sounds more filthy. Attack and launch are envelope properties that determine how fast the envelope reacts to volume changes in the input signal. Together, they form the dynamic nature of distortions. Note that if the envelope is set to zero, they will have no effect. Cut or enhance the final signal level of the device using the output disk. Zigzag can be reduced by enabling high quality mode, which can be accessed by right-click (PC) /CTRL-click (Mac) context menu. This improves sound quality, especially with high frequency signals, but there is a slight increase in CPU use.22.13 Echo effect. (Note: Echo effect is not available in the introduced versions, Lite and Metrics.) Echo is sound effect external sound effect. (Note: The external audio effect is not available in the introduced versions and Lite.) The external sound effect is slightly different from other live effects devices. Instead of processing the sound itself, it allows you to use external effects processors (devices) within the track hardware series. Choose audio outputs on your computer's audio that will go to the external device, which the audio chooses which chooses input that will bring the processing signal back to Live. As with input and track outputs, the list of available inputs and outputs depends on audio preferences, which can be accessed by configuring... Option at the bottom of each picker. Below each picker is a peak level indicator that shows the highest volume achieved. Click on the pointers to reset them. The gain handles next to the pickers will adjust the levels out of and back to Live. These levels must be carefully set to avoid a shot, either in the external device or when the sound is returned to your computer. Dry/wet control adjusts the balance between processed and dry signals. Set it to 100 percent if the external sound effect is used in a return path. The reverse-phase button of the processed signal that returns to Live.Since the effects of devices that offer latency that can't detect Live automatically, you can manually compensate for any delay by adjusting the device's latency slide. The button next to this slider allows you to set the latetime compensation amount either in milliseconds or in samples. If your external device connects to Live via a digital connection, you'll want to adjust the time settings in the samples, ensuring that you keep the number of samples you select even when you change Rate. If your external device is connected to Live via a representative connection, you'll want to adjust the latency settings in milliseconds, ensuring that you keep the amount of time you select when you change the sample rate. Note that adjusting samples gives you better control, so even in cases where you work with analog devices, you may want to adjust the time in the samples in order to achieve the lowest possible arrival time. In this case, be sure to switch back to milliseconds before changing the sample rate. Note: If the delay compensation option (see 17.5) is not specified in the options list, the hardware latency slide is disabled. For instructions on how to set up a precise latetime compensation for devices, please review the driver error compensation lesson. (Note: Filter delay effect is not available in the introduced versions and Lite.) Filter delay provides three separate delay lines, each preceded by associated lowpass and highpass filters. This allows the delay to be applied only to input signal frequencies, as determined by filter settings. Notes from each of the three delays are looped back to their own input. Each of the three delays can be turned on and off independently. The candidate delay device sets delay 1 to the left channel for the input signal, delay 2 to the right and left channels, and delay 3 to the right channel. Controls on the left can exceed the outputs of the delays channels; Each delay channel filter has an associated playkey, located to the left of each X-Y controller. X-Y controllers adjust lowpass and highpass filters simultaneously for each delay. To release the filter bandwidth, click and drag on the vertical axis; To indicate the delay time to the frequency of the song, activate the sync key, which allows the partition picker to be used in the delay time. Numbered switches represent time delay in the 16 notes. For example, selecting 4 delays the signal with four 16 notes, which is equal to one pulse (quarter of a note) of delay. With active synchronization mode, the value of the time delay field changes the shortening percentage and extends the delays by fractions, thus producing the alternative type of timing effect found in drum machines. If the sync key is turned off, the delay time returns to milliseconds. In this case, to free up the delay time, click and drag up or down in the delay time field, or click in the field and write a value. The note parameter sets the amount of the output signal that is due to the input of each delay line. Very high values can lead to runaway reactions and produce high vibrations - watch your ears and speakers if you decide to check extreme comments settings! Each delay channel has its own volume control, which can run up to +12 dB to compensate for the radical filtering of the input. Dry control The level of the signal is not processed. Set it to a minimum if you use a delay in the FlangerThe Flanger effect track. (Note: The Flanger effect is not available in the Light version.) Flanger uses two parallel-time-modulated delays to create flickering effects. Flanger delays can be adjusted while controlling the delay time. The feedback control sends part of the output signal back by inserting the device, while the Polarity key (+or -) sets the polarity. Delay time and notes can be changed simultaneously using the X-Y effect controller. It is possible to periodically control the delay time using the envelope section. You can increase or decrease the amount of the envelope (or reverse its shape with negative values), and then use attack and Release controls to define the shape of the envelope. Flanger contains two LFOs to adjust the delay time for the left and right stereo channels. The LFOs have six possible waveforms: pocket, square, triangle, sawtooth up, sawtooth down and random. The extent to which LFO affects the delay is determined with the amount controlled. LFO speed is controlled with price control, which can be set in terms of Hertz. The frequency of a song can also be synchronized and set in meter divisions (for example, sixteenotes). The stage control imparts the stereo's acoustic motion by setting LFOs to run at the same frequency, but compensating for their wavelength for each other. Set this to 180, LFOs will be completely out of phase (180 degrees apart), so that when they reach their peak, the other is minimal. Spin spends two LFO speeds relative to each other. Each delay is adjusted at a different frequency, as determined by the amount of spin. Adjusting HiPass control will cut the low frequencies from the late signal. Dry/wet control adjusts the balance between treated and dry signals. Set it to 100 percent if the fanger is used in the return path. Hello quality mode can be switched off or turned off by right-click (PC) /CTRL-click (Mac) menu entry context. Hi-quality mode results in a brighter sound, but there is a very slight increase in CPU usage.22.20 Scheffer frequency shifter frequency shifter frequency shifter frequency. (Note: The frequency shift effect is not available in the introduction and light versions.) The frequency shifter moves the incoming sound frequencies up or down by a quantity specified by the user in Hertz. A small amount of transposition can lead to the effects of insults or the effects of a lullaby, while larger shifts can create rare metal sounds. The rough handles and Fine set the amount of shift that will be applied to the input. For example, if the input is a sin-in wave at 440 Hz and the frequency is adjusted to 10 Hz, the output will be a sinus wave at 540 Hz.By change mode from Shift to Ring, and shift frequency switches from classic frequencies to liquid adjustment. In loop mode, the specified frequency amount is added to and subtracted from the entry. For example, if the incoming sound signal (A) is a sinus wave at 440 Hz and the frequency is set at 100 Hz (B), the output Contains particles at 340 Hz (A-B) and 540 Hz (A+B). The drive button allows a distortion effect, while the slider below controls the distortion level. The drive is only available in loop mode. Enabling the broadband button creates a stereo effect by reversing the polarity of spread value for the correct channel. This means that increasing the propagation value will shift the frequency down in the right channel while turning it upward suprate in the left. Note that Wide does not have an effect if the spread value is set to 0.Frequency Shifter contains two LFOs to modify the frequency of the left and right stereo channels. The LFOs have six possible waveforms: pocket, square, triangle, sawtooth up, sawtooth down and random. The extent to which LFO affects frequency is set with Amount control. LFO speed is controlled with price control, which can be set in terms of Hertz. The frequency of a song can also be synchronized and set in meter divisions (for example, sixteenotes). The stage control imparts the stereo's acoustic motion by setting LFOs to run at the same frequency, but compensating for their wavelength for each other. Set this to 180, LFOs will be completely out of phase (180 degrees apart), so that when they reach their peak, the other is minimal. Spin spends two LFO speeds relative to each other. Each stereo channel is adjusted at a different frequency, as determined by the amount of spin. When using Random waveform, phase controls and religiosity are irrelevant and do not affect sound. Dry/wet control adjusts the balance between treated and dry signals. This knob is called Mix when the drive is enabled. Note that the drive effect is post-mix, which means that you can use the Shifter frequency as a pure distortion effect by enabling the drive and setting the mix to 0%. The frequency transmission is accomplished simply by adding or subtracting a value in Hertz to the incoming sound. This is different from the pitch conversion, where incoming frequency ratios (and therefore their harmonic relationships) are maintained. For example, imagine that you have an incoming sound signal consisting of octave sinus waves apart at 440 Hz and 880 Hz. To turn this pitch up octave, we multiply these frequencies by two, resulting in new frequencies at 880 Hz and 1760 Hz.22.20.1 frequency shift shift shift and sound modulation can produce some very interesting sounds. Here are some tips for using a frequency-sweled device. The tuning drum can be acoustic drums that have been sampled deceptively. The use of sample switching controls often alters the character of sounds in unrealistic ways, resulting in pinched or tubular samples. Changing frequency can be a useful alternative. Try using the device in Shift mode with a drywet amount at 100%. Then adjust the fine frequency no more than about 100 Hz up or down. This should clearly change the size and adjust from the drum while maintaining the original sample quality. Phasingto create fertile phased effects, Using very small amounts of transformation (no more than about 2 Hz). Note that the gradient is caused by the reaction of the treated and dry signals, so you won't hear any effect until you adjust the drywet balance so that both are audible; Tremoloin loop mode, frequencies under audible range (about 20 Hz) create a tremolo effect. You can also convey the feeling of stereo motion to tremolo by running wide and using small spread values. Learn more... Try putting the spectrum device (see 22.34) after the frequency changer to see how the signal changes while changing parameters. For a good overview of what's happening, try using a simple and continuous sinus wave as input.22.21 gate effect portal. (Note: The gate effect is not available in the Lite version.) The portal effect passes only signals that exceed a user-defined threshold. A gate can eliminate low-level noise that occurs between sounds (for example, hess or hum), or form a sound by turning up the threshold so that it interrupts the frequency, delays the tails, or cuts off the natural decay of the tool. As of Live 9, the portal's internal behavior has been skillfully improved. Although it works more properly than in previous live versions, Live Sets that use Gate and which were made before Live 9 may look a little different. To ensure that older sets sound the same, the gate heritage mode option will be enabled by default when you download an old set that uses the portal. You can disable this by right-clicking (PC) / CTRL-clicking (Mac) entering the context menu in the gate address bar. The display area of the gate shows the input signal level in light gray and the output signal level in a darker gray with a white outline. This allows you to see how much gates occur at any moment, and helps you set the right parameters. The threshold handle determines the sensitivity of the gate. The Threshold value in the graph is represented as a horizontal blue line, which can also be withdrawn. Return (also known as strassis) determines the difference between the level that opens the gate and the level that closes it. High hysteria values reduces the chatter caused by the gate to open and close quickly when the input signal is near the threshold level. The return value in the view is represented as an additional horizontal orange line. With the face button enabled, the gate works in the opposite direction. The signal will pass only if its level is below the threshold. The gate can interact with the input signal only as soon as it occurs. Since it also needs to apply an attack/release envelope, the gates are always a little too late. A digital portal can solve this problem simply by delaying the input signal a little bit. The gate offers three different Lookahead times: zero milliseconds, one millisecond and ten milliseconds. The results may look very different depending on this setting. The attack time determines how long it takes for the gate to switch from closed to open when the signal moves from the bottom to the Threshold. Very short attack times can produce sharp clicking sounds, while relieving long times of sound attack. When the signal moves from top to bottom of the threshold, the waiting time begins. After the waiting period expires, the gate closes within a time period set by the Release parameter. The knob determines how much attenuation will apply when the gate is closed. If it is set to -inf dB, the closed gate will mute the input signal. Setting 0.00 dB means that even if the gate is closed, there is no effect on the signal. The settings between these two extremes reduce the input to a greater or lesser degree when the gate is closed. Typically, the signal that is gated and the input source that launches the gate is the same signal. But using sidechaining, it is possible to gate the signal based on another signal level. To access sidechain parameters, the gate window unfolds by switching the button in the address bar. Enabling this section with the Sidechain button allows you to select another path from the selectors below. This makes the specified route signal work as a gate operator, rather than the signal that is already being installed. The gain handle adjusts the level of external sidechain input, while the dry/wet handle allows you to use a combination of sidechain and the original signal as the gate trigger. With dry/wet at 100%, the gate is completely operated by a sidechain source. At 0%, the side chain is effectively exceeded. Note that increased profit does not increase the size of the source signal in the mix. The sidechain sound is just the trigger for the gate and has not actually been heard. The sidechain can be used to overlay rhythmic patterns from one source to another. For example, the sound of the plate held with the rhythm of the drum loop can be played by inserting a gate on the plate track and choosing the drum loop track as a sidechain entrance. To the right of the outer section are the controls of the sidechain EQ. Enabling this section leads to the gate being run by a certain range of frequencies, rather than a full signal. These can be either frequencies in the signal to be gated or, using EQ in conjunction with an external side, frequencies in the sound of another path. The headphone button between the external sections and EQ allows you to listen to only sidechain inputs, bypassing the gate exit. Since the sidechain sound is not fed to the output, and is only the trigger for the gate, this temporary listening option can make it much easier to set sidechain parameters and hear what makes the gate work. When this button is on, the display area shows the input signal level in the side button in the green.22.22 glue CompressorThe glue compressor effect. (Note: Glue compressor effect is not available in the introduction and light versions.) Glue compressor is an analog-style compressor created in collaboration with Cytomic, based on the classic bus compressor of the famous Mixing console. Like the original Live compressor (see 22.8), the glue compressor can be used to control the basic dynamics of individual tracks, but it is mainly designed for use on the main track or group track to glue multiple sources together in a mix.The knit sill of the knob where the pressure begins. Signals that exceed the threshold are reduced by a specified amount by the ratio parameter, which determines the ratio between the I-output signal and the output. Unlike the compressor, the glue compressor does not have an adjustable knee by the user. Instead, the knee becomes sharper as the ratio increases. The attack handle values in milliseconds. The version determines how long it takes for the compressor to return to normal operation after the signal drops below the threshold. Editing handle values in seconds. When the version is set to A (automatic), the release time will be adjusted automatically based on the incoming sound signal. Auto release is actually used in glue compressor twice - one slow as a base compression value, and quick to respond to transients in signal. The automatic version may be too slow to respond to sudden changes in level, but overall it's a useful way to tame a wide range of materials in a nice way. Dry/wet adjusts the balance between compressed and uncompressed signals. At only 100% a compressed signal is heard, while only 0%, the device is effectively overridden. Another way to control the amount of pressure is with the Range slider, which determines how much compression can occur. Values between -60 and -70 dB mimic the original hardware, while values between -40 and -15 dB can be useful as an alternative to dry/wet control. At dB 0, no pressure occurs. Makeup applies to signal gain, allowing you to compensate for a drop in the level caused by compression. The value of makeup that corresponds almost to the position of the needle in the screen should lead to a level close to what you had before compression. Switch the soft clip switch waveshaper constant, useful for tame transients very loudly. When enabled, the maximum output level for the glue compressor is -5 dB. (Note that with the enabling overwork, too high peaks may still exceed 0 decibels.) The soft clip is not a transparent selector, and will distort your signal when it is active. We recommend leaving it disabled unless this type of colored distortion is what you're looking for. The needle screen glue compressor shows the amount of low gain in dB. The Led device turns into a red clip device if the device output level exceeds 0 decibels. If soft pieces are enabled, this LED turns yellow to indicate that the peaks are cut. The glue compressor with Sidechain Section.Normally, the signal is compressed and the input source that launches the compressor is the same signal. But using sidechaining, it is possible to compress the signal on the basis of another level or a specific frequency component. To access sidechain parameters, the glue compressor window unfolds by switching the button in the address bar. The side chain parameters are divided into two parts. On the left are external controls. Enabling this section with the Sidechain button allows you to select any of the live internal routing points from the selectors below. This specific source leads to work as a glue compressor operator instead of a signal that is actually compressed. The gain handle adjusts the level of external sidechain input, while the drywet handle allows you to use a combination of sidechain and the original signal as trigger glue compressor. With dry/wet at 100%, the glue compressor is completely operated by a sidechain source. At 0%, the side chain is effectively exceeded. Note that increased profit does not increase the size of the source signal in the mix. The sidechain sound is only a trigger for the glue compressor and has not actually been heard. To the right of the outer section are the controls of the sidechain EQ. Enabling this section leads to the glue compressor being operated by a certain range of frequencies, rather than a complete signal. These can be either frequencies in the compressed signal or, using EQ in conjunction with an external side chain, frequencies in the sound of another path. The headphone button between the outer sections and EQ allows you to listen to side chain input only, bypassing the glue compressor output. Since the sidechain sound is not fed to the output, and is only the trigger for the glue compressor, this temporary listening option can make it much easier to set sidechain parameters and hear what glue compressor is actually working. The specialty can be switched off by right-clicking (PC) / CTRL-click (Mac) context menu. Enabling this option leads to the glue compressor to process internally at twice the current sampling rate, which may reduce zigzag and transient cruelty. There is a slight increase in CPU use with competence enabled. Note that with the enables to exceed seal size, the level may exceed 0 decibels even with a soft clip enabled.22.23 Delayed Grain Suppall effect grain. (Note: Grain delay effect is not available in the Lite version.) The effect of delayed grains is based on the input signal to small molecules (called grains) that are delayed individually and can also have different pitches compared to the original signal source. Random pitch and time delay can create complex masses of sound and rhythm that seem to carry a small relationship to the source. This can be very useful in creating new sounds and textures, as well as getting rid of unwelcome house guests, or terrifying little pets (just kidding!). You can direct each parameter to the vertical or horizontal axis of the X-Y controller. To set a parameter to the X axis, choose it from the parameter row below the console. To set a parameter with the Y axis, use the parameter row on Side. To indicate the delay time to the frequency of the song, activate the sync key, which allows the partition picker to be used in the delay time. Numbered switches represent time delay in the 16 notes. For example, selecting 4 delays the signal with four 16 notes, which is equal to one pulse (quarter of a note) of delay. With active synchronization mode, the value of the time delay field changes the shortening percentage and extends the delays by fractions, thus producing the alternative type of timing effect found in drum machines. If the sync key is turned off, the delay time returns to milliseconds. In this case, to free up the delay time, click and drag up or down in the delay time field, or click in the field and write a value. The delay time can also be routed to the horizontal axis of the XY controller. The Spray control adds random changes in the delay time. Low values smudge the signal over time, adding noise to the sound. High spray values completely break down the source signal structure, introducing varying degrees of rhythmic chaos. This is the recommended setting for anarchists. The size and duration of each pill is a function of the frequency parameter. The sound of the pitch and the spray depends a lot on this parameter. You can shift the grain score with the Pitch parameter, which works like a rough pitch mutant. Random pitch controls add random variations to each grain. Low values create a kind of mutant chorus effect, while high values can make the original source pitch completely incomprehensible. This parameter can interact with the main control of the pitch, allowing degrees of stability and instability in the pitch structure in the sound. The note parameter sets the amount of the output signal that is due to the delay line entry. High values can lead to runaway reactions and produce high vibrations - watch your ears and your speakers if you decide to check extreme comments settings! Grain delay also has a dry/wet control, which can be directed to the vertical axis of the Y-X.22.24 Limiter Limiter Effect controller. (Note: A specific effect is not available in the Lite version.) A specific effect is a high-quality dynamic range processor that ensures that output does not exceed a certain level. The selector is ideal for use in the main track, to prevent a snapshot. The selector is basically a compressor with an infinite ratio. (For more information on compression theory, see compressor manual input (see 22.5).) The gain handle allows you to enhance or reduce the incoming level before applying the limit. The ceiling determines the absolute maximum level that the selector will output. If your incoming signal level has no peaks above the ceiling, the selector will have no effect. The Stereo/L/R key determines how a peak selector is treated that occurs on only one side of the stereo signal. In L/R mode, function selectors act as separate selectors, with a separate restriction for each channel. Stereo mode applies to both channels Neither has a climax that requires suppression. L/R mode allows the selector to apply more pressure, but with some distortion of stereo image. The Lookahead selector affects how quickly a selector responds to peaks that require compression. Shorter look times allow for more pressure, but with an increase in distortion - especially in bass. The editing handle adjusts how long it takes for a selector to return to normal operation after the signal falls under the ceiling. With Auto enabled, the data seter automatically analyzes the incoming signal and determines the appropriate release time. The counter gives a visual indication of how much of a reduction in the gain that is applied to the signal. Note that any channel faders or devices that appear after a specific may add a profit. To ensure that your final output won't clip, place the selector as the last device in the main track device series and keep your master's fader below the looperThe 0B.22.25 effect. (Note: The Looper effect is not available in the Lite version.) Looper is an audio effect based on classic real-time loop devices. Allows you to record and repeat audio, creating endless overdubs that are synced with your set. If the set is not running, Lauper can analyze the incoming sound and set the live frequency to match it. You can also pre-select the length of the loop before recording and the live pace will adapt so that your loop fits in a specified number of bars. Moreover, audio can be imported into the looper to create a wallpaper for newly overdubbed materials, or exported from the lauper as a new cutter. The top half of the Looper interface is a large viewing area optimized for easy reading during performance. During recording, the entire viewing area turns red. After recording, the screen shows the current position in the loop and the overall length of the loop in bars and chimes. Looper's transport controls work in a similar way to other Live transport controls. The button records the incoming audio record until another button is pressed. This is above any sound currently stored in Looper. Overdub continues to add additional layers of incoming sound that are the length of the material originally recorded. The play button returns the current status of looper's buffer without registering any new material. Turn off the stop button. The behavior of the transport controls changes based on whether or not Live is played. With the transfer on, Lauper acts like a clip, and is subject to a muzzle release as determined by the muzzle picker (see 4.11). When Live is turned off, the Looper transfer engages immediately, regardless of the quantity muzzle setting. Erases the clear editing button for looper buffer. If you press Clear in Overdub mode while the transfer is running, the contents of the buffer are cleared but the frequency and length are retained. Pressing Clear in any other mode resets the Tempo and length. The undo button erases everything you've done excessively since the last overdub Your original registration is retained and anything that was overridden in a previous pass. After pressing the dip, the button changes to replay, which replaces the material that was removed by the last dip. The large button below the transport controls is the multi-purpose transport button. As with normal transport buttons, the behavior of this button changes based on looper's current status and whether or not Live is running. When the buffer is empty, one click starts recording. If looper is recording, overdubbing or stopping, switch with one click to play mode. During playback, the click turns overdub mode, allowing you to switch back and forth between overdub and play over one additional click. Quickly press the button twice to stop the looper, either from playing or overdub mode. Clicking and holding the button for two seconds while on the run activates the undoing or replay. Press and press for two seconds while stopping scanning looper buffer. The scheme for multipurpose Loop transport Button BehaviorLooper's uses the multipurpose transport button for use with a midi boiler. To set Footswitch, enter MIDI map mode (see 27.2.1), click the button and press the attached footswitch. Then exit midi map Mode.The Tempo Control chooses affects how looper determines the frequency of recorded material: none: the internal looper pace is independent of the global live pace. Follow the pace of the song: The running speed of Lauper will be adjusted so that the recorded material plays back in the universal rhythm of Live. Set and follow the pace of the song: Live's global pace will be adjusted to match the frequency of the material recorded in Looper. Any subsequent changes to Live Global's pace will adjust the run speed of Looper so that the recorded material plays back at a new global pace. The log length selector is used to set the length of the recorded material. Their behavior changes depending on whether the Global Transfer is run for Live or not, and depending on the tempo control selector setting. If Live is turned off, you can set the global rhythm of Live:Song running: if the log length selector in Looper is set to the default x bars, looper will be recorded until you press another transfer button. If you select a fixed number of tapes to register by selecting another option in the picker, Lauper will record the specified time and then switch to Run or Overdub, as determined by the button next to the box. Song doesn't work: If the looper record length selector is set to default x bars, Looper will make a guess about the frequency of the material you've recorded by simply clicking on Overdub, Play or Stop. But this can happen to the pace of this twice or half as fast as you want. If you first select a fixed number of bars, the pace of The Looper will be adjusted so that your registration is appropriate at this time. The Song Control Selector determines how Looper's transport controls will affect Live:None's global transport, meaning looper's transport controls have no effect on Live's transport. Start Song will start World Live Transfer whenever looper enters play or Overdub mode. The Stop button for Looper has no effect on public transport. Start and stop song Locks Live World Transport controls for Looper Transport. Entering play or Overdub mode will start live transfer, while pressing the Stop Looper button will stop the Live transfer. Live Transport via Looper automatically starts adjusting the operating position of any applications that are connected via Ableton Link (see 30.1). This ensures that these applications still have the pace of synchronization, and also in the correct position in the musical phrase. The *2 button doubles the length of looper's recording buffer. Any material you have already registered will simply be repeated. This allows you, for example, to record a series of single bar ideas, and then overlay a series of two bar ideas. The length and frequency of looper buffer is displayed in the display area. Similarly, the button:2 cuts the length of the current buffer in half. The material is kept in the current half of the gameplay, while the other half is ignored. Let you pull me out! On the screen, looper buffer is exported as a new audio file. You can drag and drop to your browser or directly to a path, and create a new clip. The newly created webe section mode will be set to replay by default (see 9.3.4). You can also drag the audio files to the Drag Me area! , which will replace the contents of looper's buffer. You can then use this bed material for more overdubs, for example. The speed handle adjusts the operating speed of the Looper (hence the pitch). Arrow buttons up and color to the left are shortcuts to lift or lower the pitch by octaves (thus doubling or reducing the operating speed). These buttons are subject to the muzzle selector setting up. Enabling the reverse button turns the pre-recorded material back. Any material that you overdub after enabling reverse will run forward. Note that the reverse disabled and then this material will be altered; The original material will play forward again, while the material that was overdubbed while the reverse was enabled will play backwards. The inclusion of the reverse button is subject to the preparation of the muzzle selector. Feedback set the amount of previously recorded signal that is fed back to Looper when overdubbing. When set to 100%, previously recorded material will not decrease in size. When set to 50%, it will be half loud with each iteration. Any changes in the amount of comments will not take effect until the next iteration. Note that the notes have no effect in playback mode; Each iteration will be in the same folder. Input ->Output Picker provides four options for monitoring looper input: the input signal is always allowed to be audible regardless of the status of the looper playback or recording. You'll usually need to always choose when using a luper effect in a single track. It never means that the input signal will not be heard. You'll usually need to Using a K-loop effect in the return path, it can be fed by transmission levels from a variety of other tracks. Rec/OVR means that the input is heard only when recording or overplaying, but not when looper is on or off. This is useful for situations in which you feed the sound to multiple tracks, each containing its own looper. If each of these Loopers is controlled with its own foot pedal, you can switch the recording and run the state while playing on the instrument, without having to worry about monitoring settings. Rec/OVR/Stop allows the input signal to be heard only when Looper is on. This is similar to Beat Repeat's (see 22.4) insertion mode, and can be used to record material that can suddenly interrupt your live play.22.25.1 Routing notes can be used both as a source and target for internal routing (see 14.6) to other tracks. This allows you, for example, to create overdubs for loopers that constantly feed back through other track devices. To set up this command: Insert a looper on the track. Record at least one passage of material in Looper.Create another audio track. At the top of the new track sound and audio to choose, select the track that has Looper.In audio from the bottom of the new track and audio to choose, select Looper Insert. Switch the monitor of this path to in. Add additional effects devices to the device series from the new track. Put the looper in Overdub mode. Looper output will now be routed through the other track device series and then back to the same, creating increasingoverdub layers with each pass.22.26 Dynamics Multiband multi-band dynamics multi-band effect. (Note: The effect of multi-band dynamics is not available in the introduced and latin versions.) Multiband Dynamics is a flexible tool for modifying the dynamic range of audio materials. Designed primarily as a mastering processor, Multiband Dynamics allows to compress up and down and expand up to three independent frequency bands, with adjustable crossover points and envelope controls for each range. Each frequency range has a higher and lower threshold, allowing two types of dynamic processing to be used simultaneously for each range.22.26.1 Dynamics Processing Theorys to understand how to use multiband Dynamics, it helps to understand the four different ways of manipulating dynamics. When we use the term pressure, we usually talk about lowering the level of signals that exceed the threshold. This is how Live's Compressor works (see 22.8), more accurately called downward pressure because it pushes loud signals down, thereby reducing dynamic range. But it is also possible to reduce the dynamic range of the signal by raising signal levels below the threshold. This less common form of pressure is called upward pressure. As you can see from this chart, use any type of compression results in a signal with a dynamic range smaller than The opposite of pressure is expansion. A typical dilator reduces signal levels that are below the threshold. This is how the Live portal works (see 22.21), more accurately called downward expansion because it pushes quiet signals down, thereby increasing the dynamic range. It is also possible to increase the dynamic range of signals by raising signal levels above the threshold. Like upward pressure, this technique is known as upward expansion and much less common. This diagram shows that either of the two types of expansion produces a signal with a greater dynamic range. The Expansion.To descending and downward suppter: descending pressure (common): Make loud signals quiet (uncommon): Make quiet, high-altitude signals expand (uncommon): Make loud signals louder, allowing multiband Dynamics for all these types of processing. In fact, because the device allows the incoming sound to be divided into three frequency bands, and each range is both upper and lower threshold, one instance of multiple dynamics can provide six types of processing dynamics simultaneously.22.26.2 interface and high button controls and low high and low band switching or off. With both bands off, the device acts as a single-band effect. In this case, the Mid controls only affect the incoming signal. Frequency passes below the high and low buttons adjust the transitions that determine the frequency bands for each band. If the low frequency is set to 500 Hz and the high frequency is set to 2000 Hz, then the low band goes from 0 Hz to 500 Hz, the medium band from 500 Hz to 2000 Hz and the high band of 2000 Hz until whether the sound card or model rate supports. Each band has tonic and solo buttons. With the activated button disabled to a certain range, its controls are exceeded by pressing/expanding and gaining. Solo band mute others. Input handles enhance or reduce the level of each range before the dynamics processing is subjected, while the output handles to the right of the screen adjust the range levels after processing. The display area provides a way to visualize the processing of your dynamics and adjust the relevant pressure and expansion behavior. For each band, the output level is represented by large bars, while the input level before processing is represented by small bars. If the processing is not applied, the input counters will be aligned with the top of the output counters. Displays sizing along the bottom of the dB screen. While adjusting the profit or dynamic processing of the band, you can see how its output changes compared to its input. As you move the mouse over the screen, the cursor will change to an arc as it passes over the edges of the blocks on the left or right side. These blocks represent signal levels below the above thresholds and below, respectively. Pull left or right on The edges of these blocks adjust the threshold level. Constantly pressing CTRL (PC) / CMD (MAC) while pulling the threshold will adjust the same threshold for all bands. Hold down ALT (PC) / ALT (Mac) to adjust the thresholds above and down simultaneously for one range. Holding down Shift while dragging left or right allows you to adjust the threshold of a single bar with a finer resolution. While the mouse is above the middle of the block, the cursor will change to an arrow up to the bottom. Click and swipe up or down to make the signal within the specified volume range higher or quieter. Constantly pressing CTRL (PC) / CMD (Mac) while dragging up or down will adjust the same block size for all bands. Hold down alt (PC) /ALT (Mac) to adjust the volumes above and lower simultaneously for a single bar. Pressing a high key while dragging up or down allows you to adjust the volume of a single band with precision. Double clicking within the area resets the volume to the default setting. Technically, reducing the volume of storage in the cluster above the above threshold applies to downward pressure, while being raised upwards applies to expansion. Similarly, reducing the volume of trading in the bloc below the bottom threshold applies to downward expansion, while being lifted upwards, with upward pressure. In all cases, you adjust the compressor or dilator ratio. Thresholds and ratios of all bands across the column can also be adjusted to the right of the screen. Switch the T, B and A buttons at the bottom right of the display area between the time width (attack and release), the lowest (threshold and ratio) and the highest (threshold and ratio) for each range. For the above thresholds, Attack determines how long it takes to reach maximum pressure or expansion once the signal exceeds the threshold, while the version determines how long it takes for the device to return to normal operation after the signal drops below the threshold. For the thresholds below, Attack determines how long it takes to reach maximum pressure or expansion once the signal drops below the threshold, while the version determines how long it takes for the device to return to normal operation after the signal rises above the threshold. With soft knee enabled, pressure begins or gradually expands as the threshold approaches. The RMS/Peak switch also affects how quickly multi-band Dynamics responds to level changes. With peak selection, the device reacts to short peaks within a signal. RMS mode leads to be less sensitive to very short peaks and start processing only when the incoming level exceeds the threshold for a little longer. The global output handle adjusts the device's overall output gains. Time control scaling duration for all attack and version controls. This allows you to keep the same relative envelope times, but make them all the same or slower. The amount handle adjusts the pressure intensity or expansion applied to all strips. At 0%, each compressor/extender has an effective Of 1, this means that it has no effect on the signal. A multidynamics device with sidechain section.normally, the signal that is processed and the input source that runs the device is the same. However, using the side line, dynamic processing can be applied to a signal based on another signal level or specific frequency component. To access sidechain parameters, the Multiband Dynamics window unfolds by switching the button in the address bar. The Sidechain enabled button allows you to select any of the live internal routing points from the selected below. This specific source causes it to function as a device operator rather than a signal that is actually processed. The gain handle adjusts the level of external side chain input, while the dry/wet knob lets you use a combination of the side chain and the original signals as the operator. With dry/wet at 100%, the machine is fully operated by a sidechain source. At 0%, the side chain is effectively exceeded. Note that increased profit does not increase the size of the source signal in the mix. The sidechain audio is only a player of the device and has not actually been heard. The headphone button allows you to listen to sideline input only, bypassing the device's output. Since the sideline sound is not fed by output, which is just a device player, this temporary listening option can make it much easier to set sidechain parameters and hear what makes the device actually work.22.26.4 Multiband Dynamics TipsMultiband Dynamics is a feature-rich and powerful device, capable of up to six independent types of simultaneous processing. Because of this, starting can be a bit scary. Here are some real-world apps to give you some ideas. Basic multi-band CompressionBy using only the upper thresholds, Multiband Dynamics can be used as a traditional down compressor. Adjust transition points to fit your audio material, then adjust the pressure down (by dragging down in the top pieces of the screen or by setting numerical ratios to values greater than 1.) De-essing to remove the cruelty caused by high-frequency content, try enabling the top range only and limiting its frequency to about 5 kHz. Then gradually adjust the threshold and ratio to apply a subtle downward pressure. Solo band may help to hear more easily results than your adjustments. Generally, de essing works best with fairly fast attack and release times. Engineers are often asked uncompressionMastering to perform miracles, such as adding punch and energy to a mix that has already been heavily compressed, and therefore has almost no remaining transients. Most of the time, these blends have also been greatly maximized, meaning that they also have no remaining leeway. Fortunately, upward expansion can sometimes help add life back to such overly scanned substances. To do this: Reject input handle to provide some additional The above thresholds for strips are lower than the highest peaks. Add a small amount of upward expansion to each range. Be careful — excessive upward expansion can make transients very noisy. Carefully adjust the attack and release times for each band. Note that, unlike typical downward pressure, very fast attack times will increase the impact of transients, while slower times lead to a more muffled sound. Note: Adding most or selected to increase earnings after returning some peaks to your material may destroy them again.22.27 Overdrive effect. (Note: Overdrive effect is not available in the introduced versions and Lite.) Overdrive is a distortion effect that praises some of the classic pedals used by guitarists. Unlike many distortion units, it can be driven extremely hard without sacrificing dynamic range. The distortion phase is preceded by a tape filter that can be controlled by the X-Y controller. To define the filter bandwidth, click and drag on the vertical axis. To set the position of the frequency band, click and drag on the horizontal axis. These parameters can also be set across slider boxes below the X-Y screen. The drive control set the amount of distortion. Note that 0% does not mean zero distortion! A tone acts as an EQ control after distortion. At the top of the values, the signal contains more frequent content. The Dynamics slider allows you to adjust the amount of pressure applied when distortion increases. In low settings, high distortions increase internal pressure and gain make-up. In higher settings, less compression is applied. Dry/wet control adjusts the balance between treated and dry signals. Set it to 100 percent if using Overdrive in the PedalThe track.22.28 pedal effect. (Note: Pedal effect is not available in the introduction, Lightroom and standard editions). The pedal is the effect of the guitar distortion. In combination with Live Tuner (see 22.35), amp (see 22.1) and cabinet (see 22.5) effects, a great pedal to handle guitar sounds. The pedal can also be used in less traditional settings, such as independent effect on vocals, synths or drums. Gain control adjusts the amount of distortion applied to the dry signal. Note that 0% does not mean zero distortion. It is recommended that the demand to gain back to 0% and increase it slowly until you get the desired output level. When placed in front of the pedal in a device chain, the utility (see 22.36) can use a gain parameter to lower the signal even further. The global output handle adjusts the device's overall output gains. You can choose between three different types of pedals, each inspired by distortion pedals with its own distinct acoustic features: Overdrive: Warm and smoothDistortion: Tight and sy aggressiveFuzz: unstable, with broken soundPedal has a three-band EQ that adjusts the sound bell after applying distortion. EQ is adaptive, which means that the amount of (or Q) increases with increased eq quantity. Bass control is the eq peak, with a center frequency of 100 Hz. This is useful for enhancing punch in bass or drum sounds, or relieving low frequencies of guitars. Mid control is a three-way EQ enhancement. The medium frequency switch sets the mid frequency and mid control range. The midfrequency frequency is the middle of the frequency band on which mid control works. The frequency range around this center value is narrower in the lower adapter setting and wider in the higher setting. This is common in guitar pedals so it is natural to make tight cuts and boosts in low frequencies, wider reductions and boosts in high frequencies. The center frequencies for switching settings are: lowest setting, placed on the left side: 500 HzMiddle setting, positioned in the center: 1 kHzHighest setting, placed on the right side: 2 kHz Triple control is EQ racks, with a cut frequency of 3.3 kHz. This is useful for removing harsh high frequencies (or enhancing them, if this is your tea cup!). Tip: To get the EQ more fine-grained after distortion, simply leave these controls in its neutral position and instead use another EQ, such as EQ Eight (see 22.14). Switch Sub Switch low shelf filter that enhances frequencies below 250 Hz. You can use this in conjunction with bass control by turning on Sub and set bass to -100%, or turn off Sub and set bass to 100%. The dry/wet slider adjusts the balance between treated and dry signals. Zigzag can be reduced by enabling high quality mode, which can be accessed by right-click (PC) /CTRL-click (Mac) context menu. This improves sound quality, especially with high frequency signals, but there is a slight increase in cpu use.22.28.1 pedal tipsPositioning pedal in the incoming signal device series will have an impact on how the distortion will respond. For example, adding a compressor before the pedal in the machine chain will give a more balanced end result. On the other hand, adding an EQ or filter with high gains and resonance settings before the pedal can result in a screaming distortion effect. Techno Kick choose a suitable kick with long decay (for example, Kit-Core 909, with back decay). Then, choose the distortion pedal, activate the sub switch, and dial in to earn to your taste. For the added whack, move the average frequency switch to the far right most and increase the mid-control. For more buzz, you can increase bass control. To reduce air, reduce triple control. Drum Fizzle set to add fizzle to drum set, choose the fluff pedal, increase gain to 50% and make sure to disable the sub switch. Reduce bass controls and medium to -100%, adjust the mid frequency switch to taste. Triple increase to 100%. Set output to -20dB. Then, turn the dry/wet slider down to 0% and slowly increase it until the drums fail to your taste. Broken head select Pedal, make sure the sub switch is disabled. Reject the bass control completely, and set the trilogy to 25%. Set the control in the middle of 100% and move the medium frequency switch to the far right most position. Finally, set the gain control to 100%. To add upper harmonics and warmth to a simple sub bass, choose the OD pedal, turn on the sub switch and turn on the bass control. Then, slowly increase gain until the desired effect is reached. You can then cut or increase the medium frequencies using phaser phaser effect control in the middle of phaser. (Note: Phaser effect is not available in the Lite version.) Phaser uses a series of filters that all pass to create a phase shift in the frequency spectrum of sound. Bipolar control creates cracks in the frequency spectrum. The control can then be used to reverse waveform notes and turn these cracks into peaks (or columns). The filter reduction frequency is changed with frequency control, which can be adjusted along with the notes using the X-Y controller for the effect. The device has two modes, space and earth, to change the spacing of cracks along the spectrum, thus the color of the sound. This effect can be adjusted with the color control. It is possible to periodically control the filter frequency using the envelope section. You can increase or decrease the amount of the envelope (or reverse its shape with negative values), and then use attack and Release controls to define the shape of the envelope. Veer contains two LFOs to modify the candidate frequency for the right and left stereo channels. The LFOs have six possible waveforms: pocket, square, triangle, sawtooth up, sawtooth down and random. The extent to which LFO affects the filter frequency is set with Amount control. LFO speed is controlled with price control, which can be set in terms of Hertz. The frequency of a song can also be synchronized and set in meter divisions (for example, sixteennotes). The stage control imparts the stereo's acoustic motion by setting LFOs to run at the same frequency, but compensating for their wavelength for each other. Set this to 180, LFOs will be completely out of phase (180 degrees apart), so that when they reach their peak, the other is minimal. Spin spends two LFO speeds relative to each other. Each filter frequency is then adjusted using a different LFO frequency, as determined by the amount of rotation. Dry/wet control adjusts the balance between treated and dry signals. Set it to 100 percent if using Phaser in track.22.30 ReduxThe ReduxE Effect.Nostalgic for the famous low-resolution audio quality of Mirage Ensoniq, Fairlight CMI or Commodore-64 PC? Redux takes us back to digital dark ages by reducing the signal sample rate and bit accuracy. The Downsample section has two parameters: Downsample and downsample mode switch. If the dial is set down to 1, each input sample passes to the output and the signal does not change. If it is set to 2, only each second sample will be So the result looks a little more digital. The higher the number, the lower the sample rate resulting, the more disassembled the sound. The shorthand is like applying a mosaic effect to an image: there is a loss of information and sharp edges between the blocks. The switch determines Downsample mode if the reduction is either adjusted on a smaller scale (soft, to 20.0 samples) or is not considered above a larger range (hard, down to 200 samples). Bit reduction is similar, but while reducing the reduction above the grid in time, the bit limit does the same for capacity. If the capacity disk is set by a bit limit to 8, the capacity levels are muzzled to 256 steps (8-bit resolution). If set to 1, the result is very brutal: each sample contains either a complete negative signal or a complete positive, with nothing between them. The bit limit determines the 0dB entry signal as 16-bit. Signals above 0dB are cut, and the red overload LED light will light up. Turn off bit-reducing results in a modest CPU saving.22.31 ResonatorsThe resonant effect. (Note: The resonance effect is not available in the rendered versions and the light.) This device consists of five parallel resins that overlay a tonal character on the input source. Sounds similar to anything from strings that he picked up can produce encrypted effects. Resonants are tuned into semis, providing a musical way to modify them. The first resonant sets the root pitch and the other four are tuned for this pitch in musical periods. The input signal passes first through a filter, then to the resonant. There are four types of input filters to choose from: lowpass, bandpass, highpass and chch. The input filter frequency can be adjusted with the frequency parameter. The first resonant is fed to the left and eni input channels, while the second and fourth resonators are allocated to the left, third and fifth channels to the right channel. The note parameter determines the root in each resonant that ranges from C-1 to C5. It can also be detuned in cents using fine parameter. The decay parameter lets you adjust the amount of time it takes for the resonance to be silent after getting the input signal. The longer the decay, the more tonal the result, similar to the behavior of an unstretched piano series. As with a real series, the decay time depends on the pitch, so low notes will last longer than the higher notes. The Const key holds a constant decay time regardless of the actual pitch. The resonant provides two different resonant modes. Mode A provides a more realistic sounding resonant, while Mode B provides an interesting effect especially when the Resonator I Note parameter is set to lower pitches. The resulting sound brightness can be adjusted using a color control. All resonants have a key to turn on/off and control gain. The turned off ringdevice does not need the CPU. Stop the first ringing. The others. The second resonant through the v follow the parameter a specific note in the first resonant, but each can have an individual +/-24 half-tone move using pitch controls and dtuned in cents using Detune controls. The output section features a mandatory dry/wet control and display parameter that only affects the wet signal and blends the left and right outputs of the V II renatores in a single signal if it is set to zero.22.32 ReverbThe reverbThe effect of the revere.22.32.1 input processed input traffic input input input first through high and low lower filters, which X-Y controller allows to change the midfrequency in the range (X-axis) and bandwidth (Y-axis). Any filters may be turned off when cpu power is not needed. Predelay controls the delay time in milliseconds before the first early reflection begins. This causes the bang to be delayed relative to the input signal. One's impression of the size of the real room depends in part on this delay. Typical values of natural sounds range from 1ms to 25ms.22.32.2 early reflectionsThese are the closest echoes to be heard after they bounce off the walls of the room, before the start of a diffuse-tailed echo. Its breadth and distribution give an impression of the character of the room. The control of the shape sculpts the emergence of early reflections, as well as overlaps with diffuse sound. With small values, reflections decompose more gradually and deployed sound occurs sooner, resulting in greater overlap between these components. With large values, reflections decompose more quickly and diffuse appear occurs later. A higher value can sometimes improve source clarity, while a lower value may give a smoother resolution. Spin modulation applies to early reflections. The X-Y control reaches the depth and frequency of these configurations. The greater depth tends to provide a less colorful late deployment response (more spectrum neutral). If the modulation frequency is too high, there will be a doppler frequency shift for the source sound, along with surreal washing effects. Spin may be turned off, using the paired key, to provide a modest CPU.22.32.3 Global settings that the quality selector controls in bartering between performance frequency quality. Eco uses minimal CPU resources, while Aly offers the richest echo. The Size parameter controls the room volume. In one extreme, very large size will lend the shift, diffuse delay effect to frequency. The other extreme -- a very small value -- will give it a high-color metallic sensation. The stereo image control determines the width of the output stereo image. At the top of 120 degrees, each ear receives a channel that resonates independently of the other (this is also a spread property in real rooms). The lowest setting blends the output signal to mono.22.32.4 network propagation network propagation network creates a resonant tail that follows early reflections. Adjust decay time control The time required for this tail to decrease to 1/1000th (-60 dB) of the initial capacity. High and low-frequency rack filters provide frequency-based oscillation scar. High frequency decay models absorb sound energy due to air, walls and other materials in the room (people, carpets and so on). Low shelf provides thinner decay. Each filter may be turned off to save CPU consumption. Freeze control freeze a diffuse response from input audio. When on, the echoes will continue almost endlessly. Cut the freeze adjustment by preventing the input signal from adding to the frozen echo; Flat exceeds high and low shelf filters when freezing on. If Flat is off, the frozen honesty will lose power in the debilitating frequency bands, depending on the condition of high and low rack filters. The echo density and scale parameters provide additional control over the density and roughness of the spread, and when the room size is too small, it has a significant impact on the coloring in which the propagation contributes. The chorus section adds a little adjustment and movement to the spread. Like spin section, you can control the modulation frequency and amplitude, or turn it off.22.32.5 Output in frequency output, you can adjust the overall mix of dry/wet to impact, change reflection capacity and spread with reflection level controls and spread.22.33 saturation effect. (Note: The saturation effect is not available in the Lite version.) Saturation is a waveshaping effect that can add that missing dirt, punch or warmth to your sound. It can coat input signals with soft saturation or push them to many different flavors of distortion. The X-Y network helps visualize the saturation formation curve. Problem i-and-output values are assigned to X and Y axes, respectively. The curve determines the transport function, the range by which output values fluctuate relative to input values. Because this process is usually non-linear, an incoming signal is reshaped to a greater or lesser degree depending on its level at every moment in time. The incoming signals are first cut to the dB level set by the Drive control. The counter on the right side of the screen shows how saturation affects the signal. Six fixed signal formation modes: analog clip, soft pocket, medium curve, steel curve, sinoid fold and digital clip. There is also a flexible Waveshaper mode, featuring six adjustable wavehaping parameters. In analog clip media and digital clipping, the signal is cut completely and instantly. Soft pocket, medium-oriented and steel curve softening signal modes cut to varying degrees. Fold sinoid mode can be good for special effects. The most dramatic effects can be created by selecting waveshaper curve, which has a special custom set of controls. To access these six tagged fields, saturation unfolds By switching the button in the address bar. The six additional parameters in Waveshaper mode are: drive, lane, curve, humidity, depth, and Period.Drive determines how the input signal is affected by waveshaper parameters. Setting up the drive to zero will completely deny the effect. Lin works with curve and depth parameters to change the linear part of the modulation curve. The curve mostly adds third-order harmonics to the input signal. The moisture flattens any signal near the origin of the grid. It's acting like a super-fast noise gate. The depth controls the capacity of a sinus wave that is imposed on the distortion curve. The period determines the density of ripples in the imposed sinus wave. The DC button activates the DC filter in the saturation input phase. This is mainly useful for removing DC displacements from the audio material it contains. Activating the color button allows filtering factories. The first of these, controlled by the control base, dictates how much the effect will be reduced or increased to very low frequencies. The second filter, which is essentially equivalent, is used to control higher frequencies. It is shaped with Freq (frequency cutting), width and depth controls. The output control softens the level when the device is ejected. When the soft clip switch is activated, you will also saturate the application of an instance of a representative clip curve on the output. Dry/wet control adjusts the balance between treated and dry signals. Set it to 100 percent when you use saturation in the return path. Zigzag can be reduced by enabling high quality mode, which can be accessed by right-click (PC) /CTRL-click (Mac) context menu. This improves sound quality, especially with high frequency signals, but there is a slight increase in cpu use.22.34 SpectrumThe spectrum device. (Note: The spectrum device is not available in the introduction and light versions.) The spectrum performs the analysis of the actual frequencies of the incoming sound signals. The results are represented in a graph, with dB along the vertical axis and the frequency/pitch along the horizontal. Peak levels are kept on the graph until the song is replayed. Note that the spectrum is not an acoustic effect, but a measuring tool - it does not change the signal received in any way. The cluster selection chooses the number of samples to be analyzed in each measurement. Higher values lead to better accuracy but at the expense of increased CPU load. The channel determines the channel that is analyzed — left, right, or both. The update slider determines how often Spectrum should perform an analysis. As with a block parameter, this allows for a trade-off between accuracy and CPU load. Fast response time is more accurate, but also more intense CPU. The Avg slider allows you to determine the number of blocks of samples that will be selected for each screen update. With one setting, each block is displayed. This leads to much more activity in width, which can be useful for finding the spectrum of short peaks. As Increase the average value, smoother screen updates, providing an average spectrum over time. This is more consistent with the way we actually hear it. The Graph button switches between spectrum width as a single line and separate frequency containers. Max replaces the width of the accumulated maximum capacity. With maximum enabled, you can reset maximum capacity by clicking on the screen. X-gauge buttons allow you to switch the frequency width measurement between linear, logarithmic, and single. Note that logarithm and high are actually the same sizing, but switch the legend at the top of the display between Hertz and note names. Linear measurement is particularly useful in detailed high frequency analysis. As you move the mouse over the Spectrum screen, a box appears that shows the capacity, frequency, and name of the note in the cursor position. The Range/Auto button at the bottom left of spectrum interface switches between adjusting the dynamic range of the screen manually and automatically. With scope selection, you can zoom in and scroll through the capacity by moving the mouse over the capacity legend on the left side of the screen. Drag vertically to scroll horizontally to zoom in/out. You can also use range slides to set the minimum and maximum capacity values shown. With auto-selected, the screen automatically scales itself based on the volume contained. Note that in automode, scale sliders and zoom are disabled. For a better view, you can switch the screen position between the device chain and the main window of Live by clicking the button in the Spectrum address bar or by double-clicking in display.22.35 Tuner Tuner Device.Tuner Device.Tuner displays the incoming monostadium plus the distance from the nearest half. Based on classic guitar synths, the large tuner display is designed for easy visibility on stage, ideal for adjusting external instruments or synthesizers. Note that the tuner is not an acoustic effect, but a measuring tool - it does not change the signal received in any way. The two buttons in the lower left switch between two of the main views of the Mont Tuner. A classic show that resembles traditional analog albums while the view of the explanatory stands shows the pitch over time. In both shows, the screen uses color to help indicate the accuracy of the settings. Green means in tune, while red means outside the melody. In the classic view, the incoming pitch is represented as a colored ball along a curve, and the nearest note name detected appears in the center of the screen. Arrows on either side of the note name light up to indicate whether the signal needs to be adjusted higher or lower in order to reach the desired pitch. In goal mode, a circular outline in the center of the curve shows the desired pitch, indicating in harmony if the colored ball is within the exact chart. If the incoming signal is sharp, the ball will appear to the right of the target, while the flat signals will appear Left. Tuner in the target Mode.In the strobe mode, the rotating band curve becomes of lights. The rotation altogether indicates that the signal is sharp or flat. If the band rotates to the right, the incoming pitch is sharp, while flat signals rotate the range to the left. The more the signal is outside the melody, the sooner the band moves. Switch the tuner in Strobe.The Hertz/Cents Switch mode between showing the absolute frequency of the signal in Hertz or the distance from the target pitch by cents. In the graph view, the pitch is displayed over time. The scale on the right of the display displays potential observation names, and horizontal gray straps ideally represent the center of the perfectly symin tones. Sharp notes will appear above the corresponding gray line, while flat notes will appear below. The tuner in View.Drag is listed in the view or down in the display to scroll to different pitches, or drag horizontally to zoom in or out. With automatic enabled, the screen will be automatically adjusted so that the pitch is in the center of the screen. The reference slider allows you to change the setting reference that the tuner uses when analyzing incoming signals. By default this is set to 440 Hz, which is a concert setting standard, but can be changed to any value between 410-480 Hz.Note: The tuner is designed to analyze monophonic pitches, and works best with a clean and clear signal. Multi-voice, noisy, or consistently rich signals may yield inaccurate results.22.36 UtilityThe Utility Effect.Utility can perform some very useful tasks, especially in combination with other devices. There are two separate stage controls, one for each input channel (left and right). As their names indicate, they reverse the stage of each channel. The channel mode selector allows selective processing of the left and left channels of the sample. If the left is selected, for example, the right channel is ignored and the left channel appears on both exits. This is especially useful if you have a stereo file that contains different information on both channels and you want to use only one. The Width control determines the stereo width of the wet signal. 0% give a single signal while values above 100% create an extended stereo panorama. Choosing a mid/side mode of right-click display control (PC) / CTRL-click (Mac) context menu allows you to switch between display and M/S balance controls. M/S Balance control acts as a continuous mono controller to stereo when set from 0 to 100 meters. Set the parameter to 100m will total the sound to mono. Values between 0 and 100S emphasize stereo components or out-of-stage signaling. In the 100s, only a side signal will be heard. The left and right channels will be 180 degrees of phase together. Note that if either left or right are selected in the channel mode selector, the Width and M/S Balance controls have no function and are therefore disabled. When the single switch is enabled, the stereo input signal is converted to mono. The Monoswitch converts low frequencies to the input signal to mono. This is useful for avoiding low frequency coloring when it is returned in mono. You can use the Bass Mono Frequency slider to adjust the cutting frequency between 50-500 Hz. When basmono testing is enabled, only low frequencies can be heard. This can be useful for adjusting bass mono Frequency.The gain control adjusts the input signal level from -infinite decibels to +35 dB. This can be particularly useful for automating fading volume on the track, while editing volume control for that track to balance the mix. (Note: When you adjust the Gain parameter between -18 and +35 dB using the arrow keys up and the color, the value increases or decreases in 1 decibel increments. However, between -18 dB and -inf dB, the value accelerates smoothly.) Control the balance of signal pans anywhere in the stereo field. Simply shut down the mute button of the incoming signal when it is enabled. Note: Active/mute controls are always placed in the path at the end of the signal chain. However, where you can put the utility anywhere in the signal chain, you can use its mute function to lower the input of a delay line or frequency without turning off the output of these devices. The DC key filters the DC offset and extremely low frequencies that are much below the audible range. It will have an acoustic effect only if the signal contains these frequencies and is processed after utility with non-linear effects such as compressors or waveshapers.22.37 vinyl distortion deformation vinyl. (Note: Vinyl distortion effect is not available in the introduction and light versions.) The vinyl distortion effect mimics some of the typical distortions that occur on vinyl records during operation. These distortions are caused by the geometric relations between the needle and the registered groove. The effect also features a crackling generator to add noisy antiques. The section adds the tracking model to the harmonic distortion to the input signal. Adjust the amount of distortion with the drive handle, or click and drag vertically in the X-Y tracking model display. To adjust the distortion frequency or color, drag horizontally in the X-Y screen or double-click freq field and type in value. The ALT (PC) /ALT (MAC) adjustment contract during vertical lycation in the X-Y display changes the bandwidth x (bandwidth). The section adds the effect of a single harmonic pinch to the input signal. These distortions usually occur 180 degrees outside the stage, creating a richer stereo image. The pinch effect has the same controls as the tracking model, but generates a somewhat different sound. The engine control increases or reduces the total distortion volume created by both the tracking model and Pinch.There are two distortion modes: soft and hard. Soft mode mimics the sound of a dub panel, while the hard mode is more like a standard vinyl record. Stereo/mono key determines whether a pinch distortion occurs in the stereo Single. Set it to stereo for realistic simulation of vinyl distortions. The crackling section adds noise to the signal, with the noise intensity set by the density control. The volume control adjusts the amount of gain applied to noise.22.38 Vocoder The Vocoder effect. (Note: Voice coded effect is not available in the introduced versions and Lite.) A voice code is an effect that combines the frequency information of the acoustic signal (called the carrier) with the amplitude of the other acoustic signal (called the changeer). The source of the changeist is generally something with a clear rhythmic character such as speech or drums, while the carrier is usually a synthesizer sound with rich harmony such as a string or pillow. The most familiar app of an audio codeist is the creation of modern synthesizerors or robotic sound effects. The sound baroness works by playing carrier and transsignals across the driveway filter banks. The output level of each of the filterisor is then analyzed and used to control the size of the filter corresponding to the carrier signal. You should insert live voice codeon on a track that contains the audio material you plan to use as your modular standard. The carrier picker then offers a variety of options for carrier signal: noise uses the internal noise generator in Vocoder as the source of the carrier. With this selector, x-Y is displayed that allows you to adjust the character of the noise. Adjusts the horizontal axis of shorthand. Click and drag to the left to reduce the sample rate from the carrier output. The vertical axis adjusts noise density. Click and drag down to reduce density. External allows you to select any internal guidance points available from the selectors below. This is the option you will want for classic Android audio applications. The clicker uses the same as a carrier. This basically output a resynthesized version of the signal-breaker, but allows you to use Vocoder in audio forming controls to adjust the sound. The tracking pitch enables a monophonic oscillator, which tunes itself to the pitch of the changer. High and low sliders allow you to reduce the frequency range that the oscillator will try to trace. Choose from a saw or one of three pulse waveforms and adjust the coarse adjusts of the oscillator over the pitch slider. Pitch tracking is particularly effective with monophonic sources such as melodic instruments or sounds. Note that the oscillator only occurs when it detects a clear degree. Then he keeps this pitch until he discovers a new one. This means that changing oscillator parameters or causing it to be reset (when assembling (see 15.3) is an encrypted audio path, for example) can cause unexpected changes in sound. With multi-voiced material or drums, pitch tracking is generally unpredictable (but can be very interesting.) especially when using external carrier sources, an ultrasonic encoded output can sometimes lose a lot of high end. Enable This results in brighter sound through spectrum normalization and carrier dynamics. The unvoiced handle adjusts the size of an additional noise generator, which is used to re-install parts of the signal of the player that is pitchless, such as f and sounds.Sens. determines the sensitivity of the unvoiced detection algorithm. At 100%, the unvoiced noise generator is always on. At 0%, only the main carrier source is used. The Fast/Slow key adjusts the speed of vocoder switching between unmissed detection and voting. Vocoder's large central area shows levels of individual band filters. Clicking inside this screen allows you to disprove these levels. The difference selection determines how many filters will be used. Using more ranges leads to more accurate analysis of the provider's frequency content, but requires more CPU. Domain scrolls adjust the frequency range that band filters will work on. For most sources, the fairly large set works well, but you may want to adjust the external boundaries if the sound becomes too miraculous or bassy. The BW control determines the bandwidth of the filters. In low ratios, each filter approaches a single frequency. As you increase bandwidth, you can increase the overlay of filter strips. Bandwidth of 100% is the most accurate, but higher or lower settings can create interesting effects. Switch the exact/retro switch between two types of filter behavior. In exact mode, all filters have the same profit and bandwidth. In retro mode, bands become narrower and louder at higher frequencies. The gate sets a threshold for bank liquidation. Any ranges whose levels below the threshold will be silent. The level slider enhances or interrupts an encrypted audio output. The depth determines how much is applied to the carrier signal from the capacity envelope in the estimate. At 0% the changed envelope is ignored. At 200%, high capacity tops will only be used. 100% results in classic vocoding. The attack controls and Release set how fast Vocoder responds to capacity changes in the variable signal. Very fast times keep the transients from the transgressor, but artifacts can cause distortion. Mono/Stereo switches determine the number of channels used for the bus and the rate. In mono mode, both the carrier and the swing are treated as monosources. Stereo uses a mono-adapter but handles the carrier in stereo. L/R handles both the conveyor and the signals of the stereo adapter. The carrier filter bank frequencies can be offset up or down by the forman handle. With the sound as a changer, formant small changes can change the virtual sex of the source. Dry/wet control adjusts the balance between treated and dry signals.22.38.1 Thibode VocoderTh section explains how to set up the most common Vocoder encrypted applications. Singing SynthesizerClassic Vocoder app is a singing synthesizer. To set this up in Live: Insert Vocoder into a track that contains your audio material. You can either. A clip that contains a pre-recorded audio clip or to process a live audio signal, connect a microphone to a channel on the audio device and choose this as the source of input (see 14.2) of the track. Insert analog-like synthesizer (see 24.1) into another path. Again, you can either create a MIDI clip to drive this synthesizer or play it live. Set the vocoder carrier selector for External.Select track synthesizer in vocoder audio from the chosen ones. (For best results, choose Post FX in Picker Down.) If you're creating synthesizers and audio materials in real time, make sure the arm button is enabled on both tracks. Play the synthesizer as you speak in the microphone. You will hear the rhythm of your speech, but with timbral character and frequencies of synthesizer. To hear the vocoded signal alone, solo audio track so that the natural synthesizer track is silent. Note: You'll generally get the best results if your synthesizer sound is bright and harmonious. Try sawtooth-based patches to improve sound clarity. For greater brightness and clarity, try adjusting unmissed control and/or enabling Enhance.Formant ShifterIf an audio coder is set to use the transator as its conveyor, which can be used as a powerful shape shifter. To do this: Set the carrier picker to Modulator.Set depth to 100%. Enable enhanced experience with different settings of the Formant handle to change the character of the source. For more audio carving capabilities, try adjusting different filter bank parameters as well. All right.