

MDrumStrip



Overview

MDrumStrip is a unique solution allowing you to quickly and easily mix modern punchy acoustic drums for pop/rock/metal and other genres. The way we learned to like mixes in the contemporary genres is a very different from the acoustic origins and how it was decades ago. Getting there is somewhat an alchemy and there's a very long learning curve to get there. And the drums play the lead, not only by being perhaps the most important element, it's also the most difficult to mix. The purpose of this plugin is to make things quick and easy. It can be exploited for electronic drums too, but it's designed specifically for acoustic drums.

MDrumStrip is a collection of devices (kind of like a collection of plugins, just accessible easier) created for each individual instrument and stage in the mix, so there is a device for snare drum, bass drum, overheads etc. Here we give you a relatively simple "algorithm" for drum mixing, whether it is recorded or sampled.

Recording acoustic drums

Before we get to the mixing, a quick elementary lesson about drum recording is probably in order. I will be describing the contemporary drum recording, not various oldschool ways, which have their own sound, but are not really usable for the music we are producing now.

A typical drumset consists of a bass drum, snare drum, toms, hihat and cymbals.

Bass drum is usually recorded using a dedicated dynamic mic placed inside the hole in the back (or even outside) and the exact location really depends on the drum and tuning. A "sub" is often added behind the drum, however I personally have mixed feelings about this. Another mic can also be placed on the side of the better to capture the "slap" in each hit.

Snare drum is usually recorded with a dynamic mic placed above its rim directed about to the center and a snare mic below, which is usually added just a bit to the mix as it is quite "rattling". Condenser mics can be used, but they tend to capture too much of the bleed from the cymbals and other drums.

Toms are usually recorded with a single mic above placed above its rim directed about to the center. Toms are especially prone to bleed from the cymbals and other drums.

Hihat is usually recorded using a single mic above it.

Overheads are always paired, one on the left side and one on the right side of the kit, usually placed quite a bit above. More than one pair is certainly useful if possible. Overheads mainly server for capturing the cymbals, however they record the toms as well and as such it is quite a good idea to test the stereo field to make sure the toms are spaced nicely from left to right, since close mics will be probably set to match

that.

Finally there are room mics, which provide most of the ambience, size and sense of space. Again paired, if at all possible. No reverb can ever mimic, what room mics do, since reverbs produce ambience of an infinitely small sound source, but the beauty is in the size of the drumkit itself and the complex reflections the room makes for them. Therefore the room the drums are recorded in is essential and provides a big part of the character. MDrumStrip can certainly help with situations where your room is not right, just know that it will never sound as good as if the drums were recorded in a great sounding space.

Finally a little note on drums. Cymbal sound can only be affected by the selection of the cymbals, that's actually important and selection of cymbals should be somewhat dependent on the drums and genre.

So the drums... When recording anything meaningful it is a good idea to use new drum heads. Differences between heads don't seem to be too big, but usually the thinner the better, since it will resonate with the drum the best. It will also "die" fastest though. In fact they can die before you end up recording the album. Unfortunately since the drying process is continual, it's hard to notice. Whichever heads you choose, the tuning is the key. And unfortunately tuning is difficult. However you may have the most expensive drums and mics in the world, but if you don't tune them well, they will sound horrible and there's no way to fix that later. In fact, the tuning and the drummer's hands (and sticks) may well be the most important factors when it comes to the final drum sound.

Top-down mixing

The natural way to mix is to start from individual elements, such as a snare drum, mix them into a drum bus, then mix that with all the vocals, bass, guitars... into the master bus. This could be called bottom-up mixing. Unfortunately this makes things rather difficult with modern music. First, it's common to perform very heavy processing and that could make things very difficult, since in each step you'd need to know "how the instrument should sound in that stage". Second, there are elements, mainly bass drum and snare drum, which need to cut through the mix at all times. While that can be accomplished with the use of some heavy sidechaining, there just is a simpler and more natural way.

Top-down mixing works the other way around - you start with the master track and in a way "premaster" your audio. Under that there's a drum bus, vocal bus, and a bus for everything else. This way you will always listen to sort of "mixed already" sound. And it lets you have a simple way to let bass drum and snare drum (and potentially something else too) cut through the mix - just make them a lot louder and let a compressor on the master track do it for you! Every time say a snare drum hits, the peak level gets a lot higher and the compressor tames it to make it similar to the loudness, but it keeps the relative level - the snare will still be louder than the rest, so it will cut through nicely.

So how should you make the master track? First there should be this important compressor, that does all the gluing and taming the snare/bass drum hits. **MTurboComp's MA2A** device seems to work well for this for example.

Next you can improve things with a multiband compressor, such as **MDynamicsMB** - just 2 bands would do, with a cross point at about 100-200Hz. This will improve the gluing, since bass drum (in band 1) and snare drum (in band 2) may have quite a different level.

Then, a dynamic EQ such as **MAutoDynamicEq's Compressor/Expander** device could help flattening the spectrum. Similarly **MSpectralDynamics' Flattening** device can do that, if you don't mind the latency. You can skip these though.

Then some equalization. You may be a master of EQ, most of use aren't. So there's a simple way - **MFreeformEqualizer** can perform a great spectrum matching. So just take a tune you like, a reference, Analyse it as "Source". And analyze a part of your tune, a chorus perhaps, click Equalize and you are done. Even better, use 2 of them, one in Mid channel mode, the other in Side. That way you can "mimic" both the center and side content making the stereo field similar, as that is also quite a difficult task otherwise. Note that you will need to reanalyse this from time to time while you are mixing, since your own sound is going to be changing a lot. It may be a good idea to set the MFreeformEqualizer(s) to Minimum phase if you are going to be doing some more recording, to minimize latency.

Finally some limiting, I'm sure you have some favourite limiter, **MLimiterX** or **MLimiterMB**. For heavier styles it is actually common to employ some clipping instead. **FireMaximizer** from our colleagues at UnitedPlugins could save the day, since it lets you blend between all these.

And you should definitely crosscheck your sound with your reference track(s). **MCompare** is the best solution for that.

That's all. That's the master track you put there every time, and it is actually (near) zero latency, so you can actually premix your track before you start recording the actual tune - let the musicians say play a chorus, give them headphones, and they can perhaps start recording with the "almost ready" mix in their ears. How about that? But first the drums...

Time to mix the drums!

MDrumStrip comes with several devices, some are very different. All of them are complex multieffects under the hood designed specifically for that instrument or stage. But they all have been designed to be as "similar" as possible so you can master them quickly. Also, usually the amount of changes needed should be fairly small as they employ various automatic correction mechanisms. In fact, after doing the "analysis" in each device, even the output loudness should be quite right.

Drum master bus

Since we are doing the top-down mixing, we should start with the **Master** device, and as you probably guessed it should be placed on the drum mix bus. And here's the first thing to do - play your drums and press the **AutoInput** button, leave it on for a few seconds and press it again. This analyse your drums and adapt the plugin to the level and spectrum of your audio and is present in ALL of the devices and it's

one of the essential beauties of using the plugin. Don't bother with Master anymore, just note that you will have to reanalyze it later since your sound is about to change quite dramatically.

Bass drum a'ka Kick

Bass drum is probably next. So mix all your mics you used for bass drum into a single bus (if you used more), ideally fix their phase with **MAutoAlign**, and put another MDrumStrip to the bass drum bus and select device, yes you guessed it, **Bass drum**. And again, play some audio, press **AutoInput** button, leave it on for a few seconds and press it again. Then notice the analyzer view, which contains several vertical lines, especially the most noticeable **light green one, the fundamental** - move it to the fundamental of your drum, usually the highest peak with the lowest frequency.

Now you should already have a decent kick sound, but there are a few quick things to notice. **Matching** knob automatically equalizes your kick to some hard-wired very well recorded dry bass drum sound. It's set to 50% by default, so it is already at work, but perhaps you want to change that, depending on your style and preferences.

You may want to enable the **Gate** to avoid bleed from other instruments (assuming this is not sampled drums, where you don't need to worry about that). And even better, the **BleedKill / BK** module above it is kind of like a gate for higher frequencies and considerably attenuates the bleed from cymbals, which is the most problematic.

It's time to notice the **list of modules** under the AutoInput button, where you can disable each by double clicking on them and change their order. That should easily tackle the common question of "should EQ be before compression or the other way around?". Note however that the defaults should again do the trick in most cases.

Let's check the **EQ**, which is very similar in most devices, just the way it works is different. **Depth** is great if you feel like "perhaps the EQ is doing too much". Then there are several bands, such as **Body, 2nd, Mid, Slap A...** These all do "something" specifically designed for bass drums. While each has several parameters, all you usually need to worry about is whether to enable them or not and potentially use the knob below to alter "how much and whether to cut or boost". At this point just click on each a few times - is it better on or off?

Some frequency is pinching your ear? That may well be some ugly resonance, some screw was loose in the drum or something. **ResoKill** module is what you want. You'll have to find the frequency that bothers you, so it's not for lazy people :)

Time for the **Dynamics**. The most important stuff to notice here is again the **Depth** and the **XY pad** which defines the nature of the kick. The dynamics section in most MDrumStrip devices works by employing 3 very specific algorithms, which focus on some common sounds used in various genres. And you use the XY pad to control how much of each you want. You can control parameters of each on the right, but that's only when you feel the need. In Bass drum device you have these:

Slap - mostly the initial transient, which should help the kick cut through the mix. You will probably want a lot of this for metal.

Comp - classic compression. Typical for pop and rock.

Squash - ultraovercompressed sound, which you mix in to make the kick long and big. It's quite useful for genres where there aren't too many bass drum hits and there's a need to fill the space in between.

Reverb, for a bass drum? Are you joking? Nope. It's not like you should drown your kick in that, but a little hint that makes it bigger and wider. It should be barely noticeable, but once you start using it, you won't stand your kick without it. A selection of algorithms that work well with bass drums is provided.

Finally the top right corner of the device, which features the few knobs, which all kinda define the output loudness, just different. First there's the **Maximizer**, a single parameter module, which is basically a limiter and lets you make your kick louder, but being a limiter it also squashes it. **Saturator** is similar, but performs saturation instead of limiting. Which is better? Depends on your taste, don't overthink it, just find what sounds better to your ears. But note that these are the exact parameters you should be playing with when listening to the mix as whole.

The last one, the **Output**, which is simply an output gain so it lets your kick go well above 0dBFS if needed. Now recall the section about top-down mixing, where the master compressor glues things up making sure the bass drum and snare drum cut through the mix, because these were louder than the rest in the first place. So, yes, there's nothing wrong about going above 0dBFS here, magic of the digital domain.

And that's all that's needed for processing bass drum. And the good news is that the remaining devices share the same principles, so it will only be easier from now on.

Snare drum

This journey will be nearly identical, so a quick recap. Put it on the snare bus, which should be a mix of all mics you used for your snare drum (some engineers process them individually. Seems unnecessarily complex to me).

- Put MDrumStrip with **Snare drum** device on it. Looks nearly identical, it's very different under the hood though.

- Perform the **AutoInput** analysis.

- Set the **Matching** to your taste.

- Set the **Fundamental / Green line in the analyzer** to the lowest high peak.

- Enable **BleedKill / BK** and **Gate** if needed.

- Enable/disable different bands in the **EQ** and choose whatever is better. Play with the gain knob of each if desired.

- Use **ResoKill** module to remove ugly resonance if needed. With snare drums it's unfortunately more common. BUT be careful, once you start hearing these resonances, you will be tempted to remove them all only to find out you just created a completely lifeless snare drum. That's partly why there are only 2 bands in resokill.

- Play with the XY pad in **Dynamics** module to the sound character you want. It's the same as in Bass drum actually, just Slap has been renamed to Pop.

- Play with the **Reverb** if needed. The reverb is almost always useful for snare drums, but note that the main ambience should come from your room mics.

- There's a new single-parameter-module **Phattener** on top of the device. Check it out, it can be quite magical.

- Play with the output loudness and character using **Maximizer, Saturator** and **Output**.

Toms

Since there are usually several toms and they should sound somewhat similar in terms of character and dynamics, they are processed in 2 steps. First there is a **Tom** device for each individual tom. It's pretty much just simplified version of what we had with Bass and Snare drums. So again, a quick recap:

- Perform the **AutoInput** analysis on some part with the tom playing.
- Set the **Matching** to your taste. Here it is set very low by default, since the toms differ a lot considering the broad range of sizes.
- Set the **Fundamental / Green line in the analyzer** to the lowest high peak.
- Enable **BleedKill / BK** and **Gate** if needed. It may actually be useful to go through your audio manually and literally silence the unused tom tracks wherever the tom is actually not playing. This will avoid most problems, since tom mics are especially prone to all the leakage, since they are literally surrounded by the actual drum kit and other toms.
- Enable/disable different bands in the **EQ** and choose whatever is better. Play with the gain knob of each if desired.
- Use **ResoKill** module to remove ugly resonance if needed. It's rarely needed in my opinion.

Once you process one tom, copy it to the others and just reanalyze and set the fundamentals appropriately. Assuming you tuned your toms well and used similar mics, that would do most of the heavy lifting.

The individual toms should then be panned and sent to a tom bus, make sure they are similar in loudness and there you need to put another MDrumStrip with **Tom bus** device. And you should be mostly familiar with this one as well. Let's do another recap anyways:

- There another **Gate** here, which may be handy if you are really struggling to get rid of the bleed.
- Play with the XY pad in **Dynamics** module to the sound character you want. It's the same as in Bass drum actually. In fact, in many heavier styles it is kind of desirable to make toms sound sort of like bass drums, just higher, so that complex drum fills employing both sound good.
- Play with the **Reverb** if needed. It's mostly useful for "emptier" music here, since the stereo part of the idea is not so relevant - the stereo is produced by the panning.
- Play with the output loudness and character using **Maximizer, Saturator** and **Output**.

HiHat

While the hihat is usually captured using the overheads, a dedicated hihat mic is often useful, since the hihat can be drowned by the remaining cymbals. Processing it using the **HiHat** device should be all too familiar by now. It's main point is cleaning the hihat mic sound and make its level more consistent. HiHat is usually panned to a position similar to where it is in the overheads.

Overheads

Each overhead pair should have its own bus you put your the MDrumStrip with **Overheads** device to. And guess what, it is nearly identical to the other devices, just works differently. When you listen to the processed and dry overheads, you may sometimes feel it's a bit "too much", but actually in the full mix that will sound good. That's one of the many complexities of the modern mixes.

One more thing to mention - there's a new module called **CymbalKill / CKILL**. Now why on earth would you want to remove cymbals from a mic bus designed for cymbals?? Well, sometimes you may decide that the cymbals are actually killing the drums a bit too much and the drums sound really good in the overheads. Quite rare, I admit, but perhaps that's exactly what will happen to you.

How much overheads to put into the mix? Well, mostly exactly as much, so that the cymbals are audible, but are not killing the mix. Which is usually quite low actually. But the devices have been designed, so that the "usual" level is default, so usually just keep your mix fader at 0dBFS.

Rooms

Each room pair should have its own bus you put your the MDrumStrip with **Room** device to. It may be useful to use **MAutoStereoFix** before MDrumStrip to stabilize the stereo field. The device is pretty much identical to the overheads device. And it's purpose is also similar. It just been tweaked for the room mics, which usually capture the entire kit, while overheads focus on the cymbals.

The **CymbalKill / CKILL** module is also quite useful with room mics. Often the rooms are crowded with cymbals, but we use the rooms to make these big phat snare and bass drums. So the module lets you attenuate the cymbals, so that the drums get more prominent and you can make them phat as they deserve.

How much rooms to put into the mix? As usual the default should be fairly good. Then increase it for more ambient sound, lower it for more punchy sound.

Parallel bus

You have now mixed all the individual drum tracks, hopefully let the MDrumStrip with Master device reanalyze and you are almost there. It's just there's something missing. Well, that's the parallel bus. Create a new bus in your DAW, put another MDrumStrip instance with **Parallel** device to it and send various mixed drum tracks to it. Which and how much? Well, mainly the drums and rooms. Then use the AutoInput and blahblah, you already know the drill.

Now the device is designed to really overprocess the sound in the parallel bus, so that on its own it will probably sound pretty horrible. Put the fader to silence and slowly start increasing until you notice it and just a tiny bit more. Then try to mute it and you'll hear the big difference. There's just a little bit of it, but it still makes a huge difference.

Final words

That's all that's needed to get a great drum sound. It's quite likely that at this point you noticed things you didn't notice before - some sort of ringing resonances, snare being killed by cymbals again, so you may need to go back for the best results and potentially tweak the advanced settings, but most of the heavy lifting should already be done.

Devices

This plugin can be thought of as a collection of plugins listed on the left side of the GUI. Select a device simply by clicking on it. Its GUI will appear on the right side. Compared to other MeldaProduction plugins it doesn't have any **edit screen**, so you cannot edit the internal functionality, which may be extremely complex. If you are interested in complex editing, please check **MXXX, the ultimate modular effect**.



Panic

Panic button resets the plugin state. You can use it to force the plugin to report latency to the host again and to avoid any audio problems. For example, some plugins, having a look-ahead feature, report the size of the look-ahead delay as latency, but it is inconvenient to do that every time the look-ahead changes as it usually causes the playback to stop. After you tweak the latency to the correct value, just click this button to sync the track in time with the others, minimizing phasing artifacts caused by the look-ahead delay mixing with undelayed audio signals in your host. It may also be necessary to restart playback in your host.

Another example is if some malfunctioning plugin generates extremely high values for the input of this plugin. A potential filter may start generating very high values as well and as a result the playback will stop. You can just click this button to reset the plugin and the playback will start again.

Settings

Settings

Settings button shows a menu with additional settings of the plugin. Here is a brief description of the separate items.

Licence manager lets you activate/deactivate the plugins and manage subscriptions. While you can simply drag & drop a licence file onto the plugin, in some cases there may be a faster way. For instance, you can enter your user account name and password and the plugin will do all the activating for you.

There are 4 groups of settings, each section has its own detailed help information: **GUI & Style** enables you to pick the GUI style for the plug-in and the main colours used for the background, the title bars of the windows and panels, the text and graphs area and the highlighting (used for enabled buttons, sliders, knobs etc).

Advanced settings configures several processing options for the plug-in.

Global system settings contains some settings for all MeldaProduction plugins. Once you change any of them, restart your DAW if needed, and it will affect all MeldaProduction plugins.

Dry/Wet affects determines, for Multiband plug-ins, which multiband parameters are affected by the Global dry/wet control.

Smart interpolation adjusts the interpolation algorithm used when changing parameter values; the higher the setting the higher the audio quality and the lower the chance of zippering noise, but more CPU will be used.



WWW

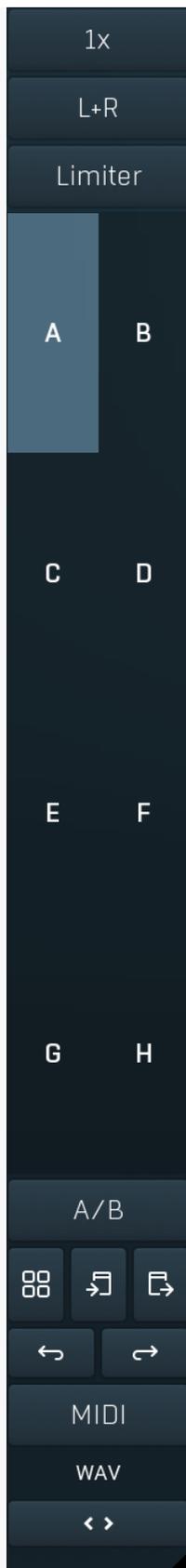
WWW button shows a menu with additional information about the plugin. You can check for updates, get easy access to support, MeldaProduction web page, video tutorials, Facebook/Twitter/YouTube channels and more.

Sleeping

Sleep indicator

Sleep indicator informs whether the plugin is currently active or in sleep mode. The plugin can automatically switch itself off to save CPU, when there is no input signal and the plugin knows it cannot produce any signal on its own and it generally makes sense. You can disable this in Settings / **Intelligent sleep on silence** both for individual instances and globally for all plugins on the system.

Plugin toolbar



Plugin toolbar provides some global features, A-H presets and more.

1x

Oversampling

Oversampling can potentially improve sound quality by processing at a higher sample rate. Processors such as compressors, saturators, distortions etc., which employ nonlinear processing generate higher harmonics of the existing frequencies. If these frequencies exceed the Nyquist rate, which equals half of the sampling rate, they get mirrored back under the Nyquist rate. This is known as aliasing and is almost always considered an artifact. This is because the mirrored frequencies are no longer harmonic and sound as digital noise as this effect does not physically occur in nature. Oversampling reduces the problem by temporarily increasing the sampling rate. This moves the Nyquist frequency which in turn, diminishes the level of the aliased harmonics. Note that the point of oversampling is not to remove harmonics, we usually add them intentionally to make the signal richer, but to reduce or attenuate the harmonics with frequencies so high, that they just cannot be represented within the sampling rate.

To understand aliasing, try this experiment: Set the sampling rate in your host to 44100 Hz. Open MOscillator and select a "rectangle" or "full saw" waveform. These simple waveforms have lots of harmonics and without oversampling even they become highly aliased. Now select 16x oversampling and listen to the difference. If you again select 1x oversampling, you can hear that the audio signal gets extensively "dirty". If you use an analyzer (MAnalyzer or MEqualizer for example), you will clearly see how, without oversampling, the plugin generates lots of inharmonic frequencies, some of them which are even below the fundamental frequency. Here is another, very extreme example to demonstrate the result of aliasing. Choose a "sine" shape and activate 16x oversampling. Now use a distortion or some saturation to process the signal. It is very probable that you will be able to hear (or at least see in the analyzer) the aliased frequencies.

The plugin implements a high-quality oversampling algorithm, which essentially works like this: First the audio material is upsampled to a higher sampling rate using a very complicated filter. It is then processed by the plugin. Further filtering is performed in order to remove any frequencies above the Nyquist rate to prevent aliasing from occurring, and then the audio gets downsampled to the original sampling rate.

Oversampling also has several disadvantages of which you should be aware before you start using it. Firstly, upsampled processing induces latency (at least in high-quality mode, although you can select low-quality directly in this popup), which is not very usable in real time applications. Secondly, oversampling also takes much more CPU power, due to both the processing being performed at a higher sampling rate (for 16x oversampling at 44100 Hz, this equates to 706 kHz!), and the complex filtering. Finally, and most importantly, oversampling creates some artifacts of its own and for some algorithms processing at higher sampling rates can actually lower the audio quality, or at least change the sound character. Your ears should always be the final judge.

As always, use this feature ONLY if you can actually hear the difference. It is a common misconception that oversampling is a miraculous cure all that makes your audio sound better. That is absolutely not the case. Ideally, you should work in a higher sampling rate (96kHz is almost always enough), while limiting the use of oversampling to some heavily distorting processors.

L+R

Channel mode

Channel mode button shows the current processing channel mode, e.g. **Left+Right (L+R)** indicates the processing of left and right channels. This is the default mode for mono and stereo audio material and effectively processes the incoming signal as expected. However the plugin also provides additional modes, of which you may take advantage as described below. Mastering this feature will give you unbelievable options for controlling the stereo field.

Note that this is not relevant for mono audio tracks, because the host supplies only one input and output channel.

Left (L) mode and Right (R) mode allow the plugin to process just one channel, only the left or only the right. This feature has a number of simple uses. Equalizing only one channel allows you to fix spectral inconsistencies, when mids are lower in one channel for example. A kind of stereo expander can be produced by equalizing each side differently. Stereo expansion could also be produced by using a modulation effect, such as a vibrato or flanger, on one of these channels. Note however that the results would not be fully mono compatible.

Left and right channels can be processed separately with different settings, by creating two instances of the plugin in series, one set to 'L' mode and the other to 'R' mode. The instance in 'L' mode will not touch the right channel and vice versa. This approach is perfectly safe and is even advantageous, as both sides can be configured completely independently with both settings visible next to each other.

Mid (M) mode allows the plugin to process the so-called mid (or mono) signal. Any stereo signal can be transformed from left and right, to mid and side, and back again, with minimal CPU usage and no loss of audio quality. The mid channel contains the mono sum (or centre), which is the signal present in both left and right channels (in phase). The side channel contains the difference between the left and right channels, which is the "stereo" part. In 'M mode' the plugin performs the conversion into mid and side channels, processes mid, leaves side intact and converts the results back into the left and right channels expected by the host.

To understand what a mid signal is, consider using a simple gain feature, available in many plugins. Setting the plugin to M mode and decreasing gain, will actually lower or attenuate the mono content and the signal will appear "wider". There must be some stereo content present, this will not work for monophonic audio material placed in stereo tracks of course. Similarly amplifying the mono content by increasing the gain, will make the mono content dominant and the stereo image will become "narrower".

As well as a simple gain control there are various creative uses for this channel mode.

Using a **compressor** on the mid channel can widen the stereo image, because in louder parts the mid part gets attenuated and the stereo becomes more prominent. This is a good trick to make the listener focus on an instrument whenever it is louder, because a wider stereo image makes the listener feel that the origin of the sound is closer to, or even around them.

A **reverb** on the mid part makes the room appear thin and distant. It is a good way to make the track wide due to the existing stereo content, yet spacey and centered at the same time. Note that since this effect does not occur naturally, the result may sound artificial on its own, however it may help you fit a dominant track into a mix.

An **equalizer** gives many possibilities - for example, the removal of frequencies that are colliding with those on another track. By processing only the mid channel you can keep the problematic frequencies in the stereo channel. This way it is possible to actually fit both tracks into the same part of the spectrum - one occupying the mid (centre) part of the signal, physically appearing further away from the listener, the other occupying the side part of the signal, appearing closer to the listener.

Using various **modulation effects** can vary the mid signal, to make the stereo signal less correlated. This creates a wider stereo image and makes the audio appear closer to the listener.

Side (S) mode is complementary to M mode, and allows processing of only the side (stereo) part of the signal leaving the mid intact. The same techniques as described for M mode can also be applied here, giving the opposite results.

Using a **gain** control with positive gain will increase the width of the stereo image.

A **compressor** can attenuate the side part in louder sections making it more monophonic and centered, placing the origin a little further away and in front of the listener.

A **reverb** may extend the stereo width and provide some natural space without affecting the mid content. This creates an interesting side-effect - the reverb gets completely cancelled out when played on a monophonic device (on a mono radio for example). With stereo

processing you have much more space to place different sounds in the mix. However when the audio is played on a monophonic system it becomes too crowded, because what was originally in two channels is now in just one and mono has a very limited capability for 2D placement. Therefore getting rid of the reverb in mono may be advantageous, because it frees some space for other instruments. An **equalizer** can amplify some frequencies in the stereo content making them more apparent and since they psycho acoustically become closer to the listener, the listener will be focused on them. Conversely, frequencies can be removed to free space for other instruments in stereo.

A **saturator / exciter** may make the stereo richer and more appealing by creating higher harmonics without affecting the mid channel, which could otherwise become crowded.

Modulation effects can achieve the same results as in mid mode, but this will vary a lot depending on the effect and the audio material. It can be used in a wide variety of creative ways.

Mid+Side (M+S) lets the plugin process both mid and side channels together using the same settings. In many cases there is no difference to L+R mode, but there are exceptions.

A **reverb** applied in M+S mode will result in minimal changes to the width of the stereo field (unless it is true-stereo, in which case mid will affect side and vice versa), it can be used therefore, to add depth without altering the width.

A **compressor** in M+S mode can be a little harder to understand. It basically stabilizes the levels of the mid and side channels. When channel linking is disabled in the compressor, you can expect some variations in the sound field, because the compressor will attenuate the louder channel (usually the mid), changing the stereo width depending on the audio level. When channel linking is enabled, a compressor will usually react similarly to the L+R channel mode.

Exciters or saturators are both nonlinear processors, their outputs depend on the level of the input, so the dominant channel (usually mid) will be saturated more. This will usually make the stereo image slightly thinner and can be used as a creative effect.

How to modify mid and side with different settings? The answer is the same as for the L and R channels. Use two instances of the plugin one after another, one in M mode, the other in S mode. The instance in M mode will not change the side channel and vice versa.

Left+Right(neg) (L+R-) mode is the same as L+R mode, but the the right channel's phase will be inverted. This may come in handy if the L and R channels seem out of phase. When used on a normal track, it will force the channels out of phase. This may sound like an extreme stereo expansion, but is usually extremely fatiguing on the ears. It is also not mono compatible - on a mono device the track will probably become almost silent. Therefore be advised to use this only if the channels are actually out of phase or if you have some creative intent.

There are also 4 subsidiary modes: **Left & zero Right (L(R0))**, **Right & zero Left (R(L0))**, **Mid & zero Side (M(S0))** and **Side & zero Mid (S(M0))**. Each of these processes one channel and silences the other.

Surround mode is not related to stereo processing but lets the plugin process up to 8 channels, depending on how many the host supplies. For VST2 plugins you have to first activate surround processing using the **Activate surround** item in the bottom. This is a global switch for all MeldaProduction plugins, which configures them to report 8in-8out capabilities to the host, on loading. It is disabled by default, because some hosts have trouble dealing with such plugins. After activation, restart your host to start using the surround capabilities of the plugins. Deactivation is done in the same way. Please note that all input and output busses will be multi-channel, that includes side-chain for example. For VST3/AU/AAX plugins the activation is not necessary.

First place the plugin on a surround track - a track that has more than 2 channels. Then select **Surround** from the plug-in's Channel Mode menu. The plugins will regard this mode as a natural extension of 2 channel processing. For example, a compressor will process each channel separately or measure the level by combining the levels of all of the inputs provided. Further surround processing properties, to enable/disable each channel or adjust its level, can be accessed via the **Surround settings** in the menu.

Ambisonics mode provides support for the modern 3D systems (mostly cinema and VR) with up to 64 channels (ambisonics 7th order). Support for this is still quite rare among the DAWs, so this needs to be activated in all DAWs using the **Activate ambisonics** item in the bottom. This is a global switch for all MeldaProduction plugins, which configures them to report 64in-64out capabilities to the host, on loading. After activation, restart your host to start using the ambisonics capabilities of the plugins. Deactivation is done in the same way. Please note that all input and output busses will be multi-channel, that includes side-chain for example.

First place the plugin on an ambisonics track, supported are all orders from 1st (4 channels) to 7th (64 channels). Then select **Ambisonics** from the plug-in's Channel Mode menu. Finally select the **Ambisonics settings** in the menu and configure the Ambisonics order and other settings if needed. The plugins will regard this mode as a natural extension of 2 channel processing. For example, a compressor will process each channel separately or measure the level by combining the levels of all of the inputs provided.

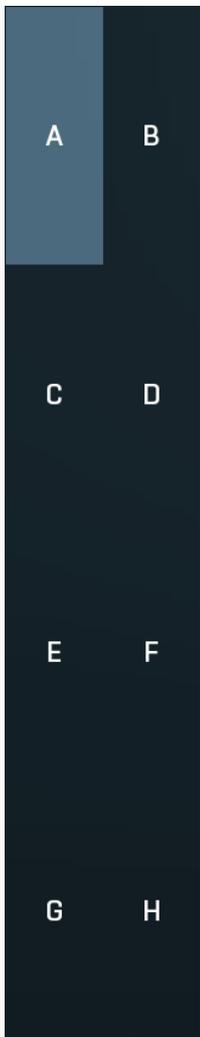
Limiters

Limiters

Limiters button enables or disables the safety limiter. Its purpose is to protect you from peaks above 0dB, which can have damaging effects to your processing chain, your monitors and even your hearing.

It is generally advised to keep your audio below 0dB at all times in all stages of your processing chain. However, several plugins may cause high level outputs with certain settings, often due to unprevented resonances with specific audio materials. The safety limiter prevents that.

Note that it is NOT wise to enable this "just in case". As with any processing, the limiter requires additional processing power and modifies the output signal. It is a transparent single-band brickwall limiter, but you still need to be careful when using it.



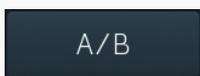
A-H presets selector

A-H presets selector controls the current A-H preset. This allows the plugin to store up to 8 sets of settings, including those parameters that cannot be automated or modulated. However it does not include channel mode, oversampling and potentially some other global controls available from the Settings/Settings menu.

For example, this feature can be used to keep multiple settings, when you are not sure about the ideal configuration. When you change any parameter, only the currently selected preset is modified.

The four buttons below enable you to switch between the last 2 selected sets using the A/B button, morph between the first 4 sets using the morphing button and copy & paste settings from one preset to another (via the clipboard).

It is also possible to switch between the presets using MIDI program change messages sent from your host. The set selected depends on the Program Change number: 0 selects A, 7 selects H, 8 selects A, 15 selects H and so on.



A/B

A/B button switches between the active and previously active A-H preset (not necessarily the A and B presets themselves). To compare any 2 of the A-H presets, select one and then the other. Clicking this button will then switch between these two. You can do the same thing by clicking on the particular presets, but this makes it easier, letting you close your eyes and just listen.



Morph

Morph button lets you morph between the A, B, C and D settings. Morphing only affects those parameters that can be automated or modulated; that does include most of the parameters however. When you click this button, an X/Y graph is shown allowing you to drag the position indicator to any position between the letters A, B, C and D. The closer you drag the indicator to one of the letters, the closer the actual settings are to that preset.

Please note that this will overwrite and change the preset that is currently selected, so it is best to select a new preset e.g. 'E', then use the morphing method. This way you will define the settings for A, B,C and D, morph between them, and store the result in 'E' without any modification of the original A, B, C and D presets.

Please note that the ABCD morphing itself cannot be automated and that, while morphing, the changes to the underlying parameters are not notified to the host (there may be hundreds of change events).



Copy

Copy button copies the current settings to the system clipboard. Other presets, oversampling, channel mode and other global settings are not copied.

Hold **Ctrl** to save the settings as a file instead. That may be necessary for complex settings, which may be too long for system clipboard to handle. It may also be advantageous when you want to send the settings via email. You can load the settings by drag & dropping them to a plugin or holding **Ctrl** and clicking **Paste**.



Paste

Paste button pastes settings from the system clipboard into the current preset. Hold **Ctrl** to load the settings from a file instead. Hold **Shift** to paste the settings to all of the A-H slots at once.



Undo

Undo button reverts the last change. Only changes to automatable or modulatable parameters and global settings (load/randomize) are stored.



Redo

Redo button reverts the last undo operation.

WAV

WAV

WAV button lets you process a file using the plugin with current settings. You can either click the button and select a file, or drag & drop the file (or multiple files) onto the button. If you let the plugin process WAV files, these will be saved with the original settings. If you use a different file type (such as MP3), the plugin will create WAV files with 32-bit bits-per-sample floating point.

Please note that the files will be overwritten, so make a copy first if you want to keep the original.



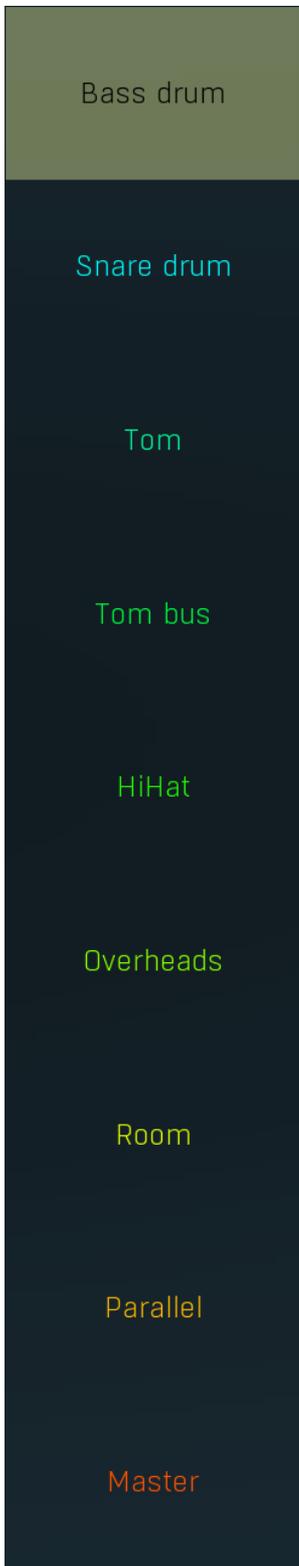
Collapse

Collapse button minimizes or enlarges the panel to release space for other editors.

Reload

Reload

Reload button reloads the device and sets all non-locked controls in the current device to their default values. It may be useful, since the plugin stores current settings when switching between the devices, hence this button is a quick and easy way to get the defaults for the devices, before you changed them. If you want to reload all parameters for the device, you must unlock the Easy screen locks or disable them all by turning off the On/Off button in the Global Locking panel in Edit mode.



Device selector

Device selector lets you choose from the predefined devices (previous 'active presets'). These are different from normal presets as they can actually have Easy-mode controls available via knobs or buttons. Click on an device to load it. Check out our video tutorials for information about creating your own devices. Although you cannot put your own devices into this selector, you can still save them as normal presets and on loading they will work in the exactly same way.

When browsing the devices, the plugin stores the control values (multiparameters). It doesn't store the full settings, only the multiparameters, so that if you switch between the devices, your settings will be kept intact, unless you switch to edit screen and perform some advanced editing, in which case it is recommended to use the A-H presets to store your work.



Collapse

Collapse button minimizes or enlarges the panel to release space for other editors.

Bass drum



Device presets

Presets

Presets button displays a window where you can load and manage available presets. Hold **Ctrl** when clicking to load a random preset instead.



Left arrow

Left arrow button loads the previous preset.



Right arrow

Right arrow button loads the next preset.



Randomize

Randomize button loads a random preset.



Matching

Matching controls the amount of automatic equalization. It should be performed on a relatively intact signal as it matches your audio to a spectrum of very good recording. Note that in some cases only a little should be applied, such as in case of overheads and rooms, where you only want to ensure there's nothing inherently "different" with your audio.

Depth

Depth controls the amount of the effect. The higher the value, the more the high frequencies are removed.

Frequency

Frequency controls the dynamic filter frequency, above which everything is attenuated unless the drum is hit. The usual setting is around 500Hz, which may seem quite brutal, but usually is surprisingly transparent and produces a solid bleed removal.

Release

Release controls the release time of the dynamic filter. The longer it is the longer it takes for the filter to start removing the high frequencies after each hit. Shorter values remove more of the bleed, but potentially also part of the sustain stage of the drum.

BK

BleedKill

BleedKill lets you remove most of the bleed from the cymbals, which are commonly obstructing drum close mics. It employs a dynamic

filter, which attenuates everything above selected frequency (mids and highs usually), but once the drum is hit, it temporarily opens and lets the attenuated frequencies pass making the drum sound as it should. This technique is much more subtle than brutal gating you'd probably use otherwise.

Low-cut

Low-cut lets you remove the lowest frequencies from the kick drum. In most cases you leave this at minimum (off), but in some music styles you may want to remove a bit of the low end to create space for bass. Also some kicks produce a bit too much ultralow frequencies, so you may use this control to tame that as well.

Fundamental frequency

Fundamental frequency controls the lowest and main frequency of the drum. This in effect controls all of the frequencies in the equalizer, so it may be considered the most important setting. In a way this defines the tuning of the drum.

You can use the green vertical line in the analyzer or control this directly. Locate the lowest high volume "bump" and set the fundamental to its center. Note that the second harmonic is sometimes louder than the fundamental. Also you may see lots of lower level "bumps" below the fundamental.

Depth

Depth controls the amount of EQ processing. The device provides a specialized equalizer for this specific instrument. If there is a **Fundamental** control, the first step would be to set it up, most likely using the analyzer. That makes the EQ tuned to that sound. The EQ then provides several aspects you can tune the sound with. These are generally specialized filters designed for this specific purpose.

EQ

EQ provides a powerful dynamic equalizer designed specifically for this particular audio material. Most devices contain the **Fundamental frequency** knob (plus accessible from the integrated analyzer), which you need to set to the lowest and usually most prominent "bump" in the spectrum and which defines the natural resonance frequency of the drum. This makes the bands of the equalizer set to proper locations (which you can further tweak using the **Shift** parameters if needed). Each band then focuses on specific part of the sound and in many cases the defaults will be all that's needed to get a great sound.

Dynamics

Dynamics controls a powerful dynamics processor, specifically designed for this particular audio material. Unlike normal compressors, which are usually a bit tricky to set up, this one contains multiple units in parallel, each focusing on specific part of the signal. While you can certainly dive into the settings themselves, usually all you need to do is to use the XY pad, which controls the mix ratio of each. For example, when processing a snare drum using the Snare device, 3 units are provided - on focusing on the classic compressor count, another on the initial "pop" used to make the snare cut through the mix, and finally the squasher, which produces very overcompressed mostly sustained sound useful to produce more body. You simply need to set the right ratio between each for your taste.

Companing

Companing controls whether the integrated compander is used and how much. Compander is a unit, which compresses the signal when reaching the dynamics section and undoes the compression afterwards. This lets the dynamics section work with more stable levels producing more consistent results. Whether that is good thing or not depends on the music style and actual sound.

Depth

Depth controls the amount of the dynamics processing. The plugin provides multiple dynamics processing units focusing on distinct characteristics of the specific drums the device is designer for, e.g. an initial transient + classic compression + squashing. You can then mix these in arbitrary ratio to get the sound you want quickly and easily. The depth then lets you mix this highly processed signal with the dry input. You can see this as parallel compression, though it is far more complex than that.

 **AUTO INPUT**

Auto-input

Auto-input is the first thing to do with the plugin. Run the audio through the plugin, click the button to enable the analysis, wait a few seconds and disable it. It will automatically adjust the internal input gain to make sure the gain staging in the plugin is ideal. If the device has **Matching** feature, this will also produce the spectrum profile for that to work.



Input

Input controls the input gain and is performed after the **Auto-input** feature. As such this is rarely useful.



Maximizer

Maximizer is a powerful limiter-based maximizer module, which lets you increase loudness of the audio, so that it penetrates the mix better. However please note that for punchy sounds (used in metal for example) the key is to get the initial transients to penetrate the mix and the maximizer may naturally tame these transients as their level is too high compared to the body of the actual drum. So in these cases use this control cautiously and potentially use **Output** control instead. That would make the output exceed 0dBFS, but that shall be fixed later when all the tracks of the drumset are mixed and glued together.



Saturator

Saturator is a powerful analogue style saturation unit, which lets you add some mojo and increase loudness on the way. Be aware that similarly to the **Maximizer** this may tame the wanted transients in more punchy styles, so use this control cautiously.



Output

Output controls the output gain. By default the output will be tuned to reach 0dBFS.

BODY

Body

Body controls the main fundamental of the kick. It can often be described as "sub" or tone of the drum.



Gain

Gain controls how much the frequencies related to this particular sound characteristic are amplified or reduced, whether statically or dynamically.

Shift

Shift lets you move actual frequency of this filter a bit. By default this is 0% meaning that the filter frequency is fully determined by the **Fundamental**. But you may alter this if the results are not meeting your expectations. The filter locations are displayed as blue lines in the analyzer.

BW

BW controls the bandwidth of the filter, in other words the range of frequencies the filter affects. If you make it very low, it will target the relevant frequencies very specifically, potentially ringing. If you make it very high, it will change a broader part of the spectrum. The filters are tuned in such a way, that the "usual optimum" is the default 50%.

Dynamic

Dynamic controls to which extent is this filter dynamic. When set to 0%, the filter becomes completely static, like if you'd use a traditional parametric equalizer. Increasing this makes the filter move with the sound. 100% makes the filter fully dynamic, meaning that when the input is silent, it has no effect, but once the level exceeds a certain threshold, the filter starts doing its job. This means that the filter will be moving only when the drum is actually hit. This is often very advantageous, since it makes the output much cleaner, not amplifying frequencies longer than actually needed.

Release

Release controls how quickly the release part of the dynamic EQ ends. The purpose of using a dynamic EQ is to change the sound ONLY during the actual hit. When a drum is hit and the main body filter is engaged, it is usually set to amplify the fundamental frequency. The main body usually ends very quickly though, but the release time can compensate for that by amplifying the frequencies a bit longer, so that the drum sounds bigger. However the release time is too long, the drum may become a bit too long, ringy and obstructing to other elements in the mix.

If the **Dynamic** is set to 0%, this control has no effect.

2nd

2nd

2nd controls the second harmonic. It is usually located around 100Hz and despite it is technically a harmonic, it is usually used to produce the body of the drum without the brutal sub the actual **Body** filter would generate.

3rd

3rd

3rd controls the third harmonic. Whether it is useful depends on the actual drum.

MID

Mid

Mid controls the middle frequencies, which are usually tamed to make the kick more "hi-fi" and to clear space for other instruments in the mix.

SLAP A

Slap A

Slap A controls the high frequencies producing the classic kick slap, which makes it cut through the mix.

SLAP B

Slap B

Slap B controls the high frequencies producing the classic kick slap, which makes it cut through the mix.

RESOKILL

Resokill

Resokill is a resonance removal module based on 2 very narrow peak filters, which let you remove unwanted resonances many drums suffer from. The knobs control the frequencies of these filters and you can also see them as red lines in the analyzer graph.



Frequency

Frequency controls the resonance frequency to be removed. You can see the location as the red line in the analyzer graph. Resonances often need to be set very accurately, so that the filter is most efficient and natural, so don't forget to use mouse-wheel or ctrl/cmd for fine-tuning.



Gain

Gain controls the gain of the resonance filter. The usual workflow is to set this to maximum, so that the filter starts amplifying the frequencies. Then use the **Frequency** knob to locate the optimal frequency where the resonance will most "unpleasant". Then lower the gain as much as needed. Note that in many cases resonances may sound ugly on their own, but they may have an important role in cutting through the mix. Therefore it is advised to use this with caution.



Harmonics

Harmonics controls to which extent are the higher harmonics removed along with the main resonance. This is often useful as natural signals are rarely just single frequencies.



BW

BW controls the bandwidth of the filter, in other words the range of frequencies the filter affects. Considering the purpose of removing unwanted resonances it is usually best to set it very low, so that it will target the relevant frequencies very specifically.



XY pad

XY pad lets you quickly set the ratio between all the provided dynamics processors.



Slap

Slap produces the initial kick transient. It is short, aggressive and potentially high in level. It lets the kick cut through the mix. The more aggressive the music is, the more of this is usually needed.



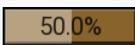
Mix

Mix controls the level of this dynamics processor in the mix.



Drive

Drive controls the aggressivity of the module. Usually the dynamics modules are tuned in such a way that the usual optimum is the default 50%.



Saturation

Saturation controls the amount of the analogue-like saturation of this dynamics processor and that gives the audio some additional "mojo" and can make it more audible in the mix. It can sound heavily distorted when overused however.



Time

Time controls the speed of the dynamics processor. Its effect depends on the specified processor, so the best way to check this is to listen to that specific processor (either by using the XY pad or by disabling the others).



Comp

Comp produces the classic kick compression. It contains the slap, but it mainly makes the kick bigger and longer.



Squash

Squash produces a brutal overcompression, which can be useful to make the kick especially long and big. It is not useful for more aggressive styles as it clutters the mix a lot in most cases.



Reverb

Reverb provides an advanced algorithmic reverb chosen specifically for processing this particular type of sound. It may be useful to make the sound wider and bigger. However note that the main effect of spaciousness should come from the room and overhead mics.

Non-linear
Room
Studio
Plate 1
Plate 2
Plate 3
Chamber
Hall

Mode

Mode lets you choose from several algorithmic reverb models specifically selected for this type of sound. The modes are usually sorted from the smallest to the biggest and most audible.



Dry/Wet

Dry/Wet controls the ratio between the dry and reverberated signals. The range is intentionally limited to 40% as that seems a useful maximum for most needs with drums.



Length

Length defines the reverberation signal length. In other words, how long the reverb will become inaudible after the drum hit. In effect this makes the virtual space smaller or bigger.

0 ms

Predelay

Predelay defines the initial delay before the actual response, which simulates the space between the sound source and the listener. The longer the predelay is, the further away the source seems. At some point (around say 100ms) the brain stops understanding that the reverberation belongs to the dry signal and starts interpreting them separately, and then the dry signal becomes close to the listener again. Predelay can therefore be used to control the distance from the source to the destination, but detaching the 2 signals can also be useful for instance to fill up a mix that isn't full enough without smearing the signal with the reverb itself.

50.00 Hz

Low-cut

Low-cut lets you remove the lowest frequencies from the reverb signal. This is often useful to produce cleaner mixes.

4000 Hz

High-cut

High-cut lets you remove the highest frequencies from the reverb signal. This is often useful to produce cleaner mixes.



Threshold

Threshold controls the lowest level at which the gate (or more specifically an expander) stops attenuating the sound. Basically, you have 2 options. A) Set the threshold low, which would mean that the gate will only attenuate if the input level is low and won't have any effect on the actual drum sound. In that case you may use high **Ratio** if needed. B) Set the threshold high, which makes the gate become more of an expander processing the overall sound and providing more dynamics. In this case lower **Ratio** is usually needed to get natural results.



Ratio

Ratio controls the downwards expander ratio and the higher it is, the more brutal the processing is. Usually the lower the **Threshold** is, the higher the ratio can be for natural results.



Release

Release controls how quickly the gate is likely to start attenuating the drum in its sustain stage. In other words, higher values makes the drum sound longer.



Gate

Gate provides a gating dynamics processor, or more specifically a downwards expander. While it can be used to reduce bleed between the drums, it's alternate purpose is to control the sustain length. For most close mics it should be placed as one of the first processors in the chain, before the dynamics gets changed, and it is used to "stop" the drum sustain at a proper time. In many cases the relatively brutal processing performed by the **Dynamics** and **Maximizer** processors may lead to the length of the drum sound to be unpleasantly long, cluttering the mix and producing an unnatural squashy sound, especially when overused for punchier music styles. The gate may then be used to lower the sustain level before it gets amplified by these processors.

Snare drum



Matching

Matching controls the amount of automatic equalization. It should be performed on a relatively intact signal as it matches your audio to a spectrum of very good recording. Note that in some cases only a little should be applied, such as in case of overheads and rooms, where you only want to ensure there's nothing inherently "different" with your audio.



Phattener

Phattener solves the problem with too short and thin snare sound. Usually you can use room mics and additional samples to do that, but the phattener solves the issue by producing a longer tail automatically, making the snare sound longer and phatter.



EQ

EQ provides a powerful dynamic equalizer designed specifically for this particular audio material. Most devices contain the **Fundamental frequency** knob (plus accessible from the integrated analyzer), which you need to set to the lowest and usually most prominent "bump" in the spectrum and which defines the natural resonance frequency of the drum. This makes the bands of the equalizer set to proper locations (which you can further tweak using the **Shift** parameters if needed). Each band then focuses on specific part of the sound and in many cases the defaults will be all that's needed to get a great sound.

DYNAMICS Dynamics

Dynamics controls a powerful dynamics processor, specifically designed for this particular audio material. Unlike normal compressors, which are usually a bit tricky to set up, this one contains multiple units in parallel, each focusing on specific part of the signal. While you can certainly dive into the settings themselves, usually all you need to do is to use the XY pad, which controls the mix ratio of each. For example, when processing a snare drum using the Snare device, 3 units are provided - on focusing on the classic compressor count, another on the initial "pop" used to make the snare cut through the mix, and finally the squasher, which produces very overcompressed mostly sustained sound useful to produce more body. You simply need to set the right ratio between each for your taste.

0.00% Companding

Companding controls whether the integrated compander is used and how much. Compander is a unit, which compresses the signal when reaching the dynamics section and undoes the compression afterwards. This lets the dynamics section work with more stable levels producing more consistent results. Whether that is good thing or not depends on the music style and actual sound.

100.0% Depth

Depth controls the amount of the dynamics processing. The plugin provides multiple dynamics processing units focusing on distinct characteristics of the specific drums the device is designer for, e.g. an initial transient + classic compression + squashing. You can then

mix these in arbitrary ratio to get the sound you want quickly and easily. The depth then lets you mix this highly processed signal with the dry input. You can see this as parallel compression, though it is far more complex than that.

AUTO
INPUT

Auto-input

Auto-input is the first thing to do with the plugin. Run the audio through the plugin, click the button to enable the analysis, wait a few seconds and disable it. It will automatically adjust the internal input gain to make sure the gain staging in the plugin is ideal. If the device has **Matching** feature, this will also produce the spectrum profile for that to work.



Input

Input controls the input gain and is performed after the **Auto-input** feature. As such this is rarely useful.



Maximizer

Maximizer is a powerful limiter-based maximizer module, which lets you increase loudness of the audio, so that it penetrates the mix better. However please note that for punchy sounds (used in metal for example) the key is to get the initial transients to penetrate the mix and the maximizer may naturally tame these transients as their level is too high compared to the body of the actual drum. So in these cases use this control cautiously and potentially use **Output** control instead. That would make the output exceed 0dBFS, but that shall be fixed later when all the tracks of the drumset are mixed and glued together.



Saturator

Saturator is a powerful analogue style saturation unit, which lets you add some mojo and increase loudness on the way. Be aware that similarly to the **Maximizer** this may tame the wanted transients in more punchy styles, so use this control cautiously.



Output

Output controls the output gain. By default the output will be tuned to reach 0dBFS.

200.0 Hz

Fundamental frequency

Fundamental frequency controls the lowest and main frequency of the drum. This in effect controls all of the frequencies in the equalizer, so it may be considered the most important setting. In a way this defines the tuning of the drum.

You can use the green vertical line in the analyzer or control this directly. Locate the lowest high volume "bump" and set the fundamental to its center. Note that the second harmonic is sometimes louder than the fundamental. Also you may see lots of lower level "bumps" below the fundamental.

100.0%

Depth

Depth controls the amount of EQ processing. The device provides a specialized equalizer for this specific instrument. If there is a **Fundamental** control, the first step would be to set it up, most likely using the analyzer. That makes the EQ tuned to that sound. The EQ then provides several aspects you can tune the sound with. These are generally specialized filters designed for this specific purpose.

BODY

Body

Body controls the main body of the snare drum. It is essential to make the drum sound big and powerful.



Gain

Gain controls how much the frequencies related to this particular sound characteristic are amplified or reduced, whether statically or dynamically.

0.00%

Shift

Shift lets you move actual frequency of this filter a bit. By default this is 0% meaning that the filter frequency is fully determined by the **Fundamental**. But you may alter this if the results are not meeting your expectations. The filter locations are displayed as blue lines in the analyzer.

50.0%

BW

BW controls the bandwidth of the filter, in other words the range of frequencies the filter affects. If you make it very low, it will target the relevant frequencies very specifically, potentially ringing. If you make it very high, it will change a broader part of the spectrum. The filters are tuned in such a way, that the "usual optimum" is the default 50%.

Off

Dynamic

Dynamic controls to which extent is this filter dynamic. When set to 0%, the filter becomes completely static, like if you'd use a traditional parametric equalizer. Increasing this makes the filter move with the sound. 100% makes the filter fully dynamic, meaning that when the input is silent, it has no effect, but once the level exceeds a certain threshold, the filter starts doing its job. This means that the filter will be moving only when the drum is actually hit. This is often very advantageous, since it makes the output much cleaner, not amplifying frequencies longer than actually needed.

200 ms

Release

Release controls how quickly the release part of the dynamic EQ ends. The purpose of using a dynamic EQ is to change the sound ONLY during the actual hit. When a drum is hit and the main body filter is engaged, it is usually set to amplify the fundamental frequency. The main body usually ends very quickly though, but the release time can compensate for that by amplifying the frequencies a bit longer, so that the drum sounds bigger. However the release time is too long, the drum may become a bit too long, ringy and obstructing to other elements in the mix.

If the **Dynamic** is set to 0%, this control has no effect.

2ND

2nd

2nd controls the second harmonic. It may let the drum sound more natural and powerful.

3RD

3rd

3rd controls the third harmonic. It may let the drum sound more natural and powerful.

4TH

4th

4th controls the 4th harmonic. It may let the drum sound more natural and powerful.

RING

Ring

Ring controls the middle frequencies, which are usually responsible for the mud and ringing.

AIR

Sparkle

Sparkle controls the highest frequencies and lets you add more air to the drum, which in effect makes it more prominent in the mix.

RESOKILL

Resokill

Resokill is a resonance removal module based on 2 very narrow peak filters, which let you remove unwanted resonances many drums suffer from. The knobs control the frequencies of these filters and you can also see them as red lines in the analyzer graph.



Frequency

Frequency controls the resonance frequency to be removed. You can see the location as the red line in the analyzer graph. Resonances often need to be set very accurately, so that the filter is most efficient and natural, so don't forget to use mouse-wheel or ctrl/cmd for fine-tuning.

0.00 dB

Gain

Gain controls the gain of the resonance filter. The usual workflow is to set this to maximum, so that the filter starts amplifying the frequencies. Then use the **Frequency** knob to locate the optimal frequency where the resonance will most "unpleasant". Then lower the gain as much as needed. Note that in many cases resonances may sound ugly on their own, but they may have an important role in cutting through the mix. Therefore it is advised to use this with caution.

0.00%

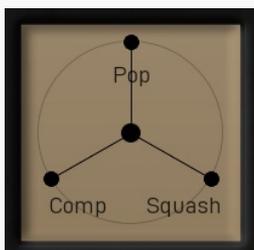
Harmonics

Harmonics controls to which extent are the higher harmonics removed along with the main resonance. This is often useful as natural signals are rarely just single frequencies.

0.00%

BW

BW controls the bandwidth of the filter, in other words the range of frequencies the filter affects. Considering the purpose of removing unwanted resonances it is usually best to set it very low, so that it will target the relevant frequencies very specifically.



XY pad

XY pad lets you quickly set the ratio between all the provided dynamics processors.

POP

Pop

Pop produces the initial snare peak. It is short, aggressive and potentially high in level. It lets the snare drum cut through the mix. The more aggressive the music is, the more of this is usually needed.



Mix

Mix controls the level of this dynamics processor in the mix.



Drive

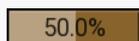
Drive controls the aggressivity of the module. Usually the dynamics modules are tuned in such a way that the usual optimum is the default 50%.



50.0%

Saturation

Saturation controls the amount of the analogue-like saturation of this dynamics processor and that gives the audio some additional "mojo" and can make it more audible in the mix. It can sound heavily distorted when overused however.



50.0%

Time

Time controls the speed of the dynamics processor. Its effect depends on the specified processor, so the best way to check this is to listen to that specific processor (either by using the XY pad or by disabling the others).



COMP

Comp

Comp produces the classic snare compression. It contains the pop, but it mainly makes the snare sound bigger and longer.



SQUASH

Squash

Squash produces a brutal overcompression, which can be useful to make the snare especially long and big. It is not useful for more aggressive styles as it clutters the mix a lot in most cases.



REVERB

Reverb

Reverb provides an advanced algorithmic reverb chosen specifically for processing this particular type of sound. It may be useful to make the sound wider and bigger. However note that the main effect of spaciousness should come from the room and overhead mics.



Non-linear

Room

Studio

Plate 1

Plate 2

Plate 3

Chamber

Hall

Mode

Mode lets you choose from several algorithmic reverb models specifically selected for this type of sound. The modes are usually sorted from the smallest to the biggest and most audible.



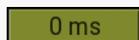
Dry/Wet

Dry/Wet controls the ratio between the dry and reverberated signals. The range is intentionally limited to 40% as that seems a useful maximum for most needs with drums.



Length

Length defines the reverberation signal length. In other words, how long the reverb will become inaudible after the drum hit. In effect this makes the virtual space smaller or bigger.



0 ms

Predelay

Predelay defines the initial delay before the actual response, which simulates the space between the sound source and the listener. The longer the predelay is, the further away the source seems. At some point (around say 100ms) the brain stops understanding that the reverberation belongs to the dry signal and starts interpreting them separately, and then the dry signal becomes close to the listener again. Predelay can therefore be used to control the distance from the source to the destination, but detaching the 2 signals can also be useful for instance to fill up a mix that isn't full enough without smearing the signal with the reverb itself.



400.0 Hz

Low-cut

Low-cut lets you remove the lowest frequencies from the reverb signal. This is often useful to produce cleaner mixes.

10000 Hz **High-cut**

High-cut lets you remove the highest frequencies from the reverb signal. This is often useful to produce cleaner mixes.

50.0% **Depth**

Depth controls the amount of the effect. The higher the value, the more the high frequencies are removed.

500.0 Hz **Frequency**

Frequency controls the dynamic filter frequency, above which everything is attenuated unless the drum is hit. The usual setting is around 500Hz, which may seem quite brutal, but usually is surprisingly transparent and produces a solid bleed removal.

100 ms **Release**

Release controls the release time of the dynamic filter. The longer it is the longer it takes for the filter to start removing the high frequencies after each hit. Shorter values remove more of the bleed, but potentially also part of the sustain stage of the drum.

BK **BleedKill**

BleedKill lets you remove most of the bleed from the cymbals, which are commonly obstructing drum close mics. It employs a dynamic filter, which attenuates everything above selected frequency (mids and highs usually), but once the drum is hit, it temporarily opens and lets the attenuated frequencies pass making the drum sound as it should. This technique is much more subtle than brutal gating you'd probably use otherwise.

-40.0 dB **Threshold**

Threshold controls the lowest level at which the gate (or more specifically an expander) stops attenuating the sound. Basically, you have 2 options. A) Set the threshold low, which would mean that the gate will only attenuate if the input level is low and won't have any effect on the actual drum sound. In that case you may use high **Ratio** if needed. B) Set the threshold high, which makes the gate become more of an expander processing the overall sound and providing more dynamics. In this case lower **Ratio** is usually needed to get natural results.

2.00 : 1 **Ratio**

Ratio controls the downwards expander ratio and the higher it is, the more brutal the processing is. Usually the lower the **Threshold** is, the higher the ratio can be for natural results.

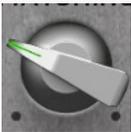
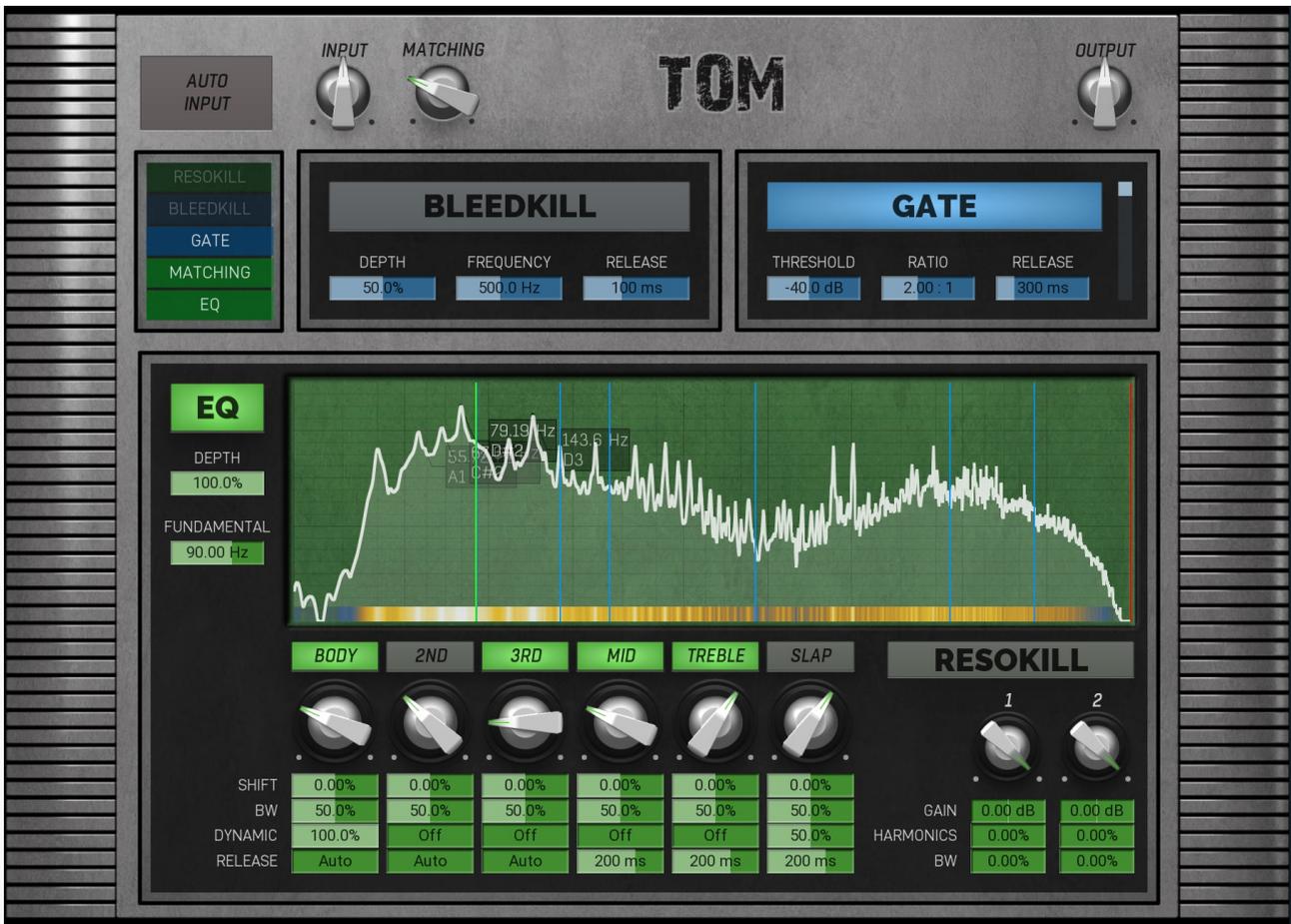
400 ms **Release**

Release controls how quickly the gate is likely to start attenuating the drum in its sustain stage. In other words, higher values makes the drum sound longer.

GATE **Gate**

Gate provides a gating dynamics processor, or more specifically a downwards expander. While it can be used to reduce bleed between the drums, it's alternate purpose is to control the sustain length. For most close mics it should be placed as one of the first processors in the chain, before the dynamics gets changed, and it is used to "stop" the drum sustain at a proper time. In many cases the relatively brutal processing performed by the **Dynamics** and **Maximizer** processors may lead to the length of the drum sound to be unpleasantly long, cluttering the mix and producing an unnatural squashy sound, especially when overused for punchier music styles. The gate may then be used to lower the sustain level before it gets amplified by these processors.

Tom



Matching

Matching controls the amount of automatic equalization. It should be performed on a relatively intact signal as it matches your audio to a spectrum of very good recording. Note that in some cases only a little should be applied, such as in case of overheads and rooms, where you only want to ensure there's nothing inherently "different" with your audio.



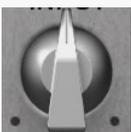
EQ

EQ provides a powerful dynamic equalizer designed specifically for this particular audio material. Most devices contain the **Fundamental frequency** knob (plus accessible from the integrated analyzer), which you need to set to the lowest and usually most prominent "bump" in the spectrum and which defines the natural resonance frequency of the drum. This makes the bands of the equalizer set to proper locations (which you can further tweak using the **Shift** parameters if needed). Each band then focuses on specific part of the sound and in many cases the defaults will be all that's needed to get a great sound.



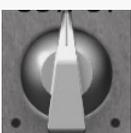
Auto-input

Auto-input is the first thing to do with the plugin. Run the audio through the plugin, click the button to enable the analysis, wait a few seconds and disable it. It will automatically adjust the internal input gain to make sure the gain staging in the plugin is ideal. If the device has **Matching** feature, this will also produce the spectrum profile for that to work.



Input

Input controls the input gain and is performed after the **Auto-input** feature. As such this is rarely useful.



Output

Output controls the output gain. By default the output will be tuned to reach 0dBFS.



Fundamental frequency

Fundamental frequency controls the lowest and main frequency of the drum. This in effect controls all of the frequencies in the equalizer, so it may be considered the most important setting. In a way this defines the tuning of the drum. You can use the green vertical line in the analyzer or control this directly. Locate the lowest high volume "bump" and set the fundamental to its center. Note that the second harmonic is sometimes louder than the fundamental. Also you may see lots of lower level "bumps" below the fundamental.

100.0%

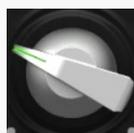
Depth

Depth controls the amount of EQ processing. The device provides a specialized equalizer for this specific instrument. If there is a **Fundamental** control, the first step would be to set it up, most likely using the analyzer. That makes the EQ tuned to that sound. The EQ then provides several aspects you can tune the sound with. These are generally specialized filters designed for this specific purpose.

BODY

Body

Body controls the main fundamental of the tom. Being part of the bass spectrum it is usually not directly amplified, as its effect would be too aggressive and clutter the mix. Also the tom sound itself usually contains plenty, so it is in fact attenuated by default.



Gain

Gain controls how much the frequencies related to this particular sound characteristic are amplified or reduced, whether statically or dynamically.

0.00%

Shift

Shift lets you move actual frequency of this filter a bit. By default this is 0% meaning that the filter frequency is fully determined by the **Fundamental**. But you may alter this if the results are not meeting your expectations. The filter locations are displayed as blue lines in the analyzer.

50.0%

BW

BW controls the bandwidth of the filter, in other words the range of frequencies the filter affects. If you make it very low, it will target the relevant frequencies very specifically, potentially ringing. If you make it very high, it will change a broader part of the spectrum. The filters are tuned in such a way, that the "usual optimum" is the default 50%.

100.0%

Dynamic

Dynamic controls to which extent is this filter dynamic. When set to 0%, the filter becomes completely static, like if you'd use a traditional parametric equalizer. Increasing this makes the filter move with the sound. 100% makes the filter fully dynamic, meaning that when the input is silent, it has no effect, but once the level exceeds a certain threshold, the filter starts doing its job. This means that the filter will be moving only when the drum is actually hit. This is often very advantageous, since it makes the output much cleaner, not amplifying frequencies longer than actually needed.

Auto

Release

Release controls how quickly the release part of the dynamic EQ ends. The purpose of using a dynamic EQ is to change the sound ONLY during the actual hit. When a drum is hit and the main body filter is engaged, it is usually set to amplify the fundamental frequency. The main body usually ends very quickly though, but the release time can compensate for that by amplifying the frequencies a bit longer, so that the drum sounds bigger. However the release time is too long, the drum may become a bit too long, ringy and obstructing to other elements in the mix.

If the **Dynamic** is set to 0%, this control has no effect.

2ND

2nd

2nd controls the second harmonic. Whether it is useful depends on the actual drum.

3RD

3rd

3rd controls the third harmonic. Whether it is useful depends on the actual drum.

MID

Mid

Mid controls the middle frequencies, which are usually tamed to make the tom more "hi-fi" and to clear space for other instruments in the mix.

TREBLE

Treble

Treble controls the high frequencies which are useful to make the tom cut through the mix and makes it sort-of hifi.

SLAP

Slep

Slep produces the classic "slap" sound, which cuts through the mix. In some genres it may be too aggressive though.

RESOKILL

Resokill

Resokill is a resonance removal module based on 2 very narrow peak filters, which let you remove unwanted resonances many drums

suffer from. The knobs control the frequencies of these filters and you can also see them as red lines in the analyzer graph.



Frequency

Frequency controls the resonance frequency to be removed. You can see the location as the red line in the analyzer graph. Resonances often need to be set very accurately, so that the filter is most efficient and natural, so don't forget to use mouse-wheel or ctrl/cmd for fine-tuning.

0.00 dB

Gain

Gain controls the gain of the resonance filter. The usual workflow is to set this to maximum, so that the filter starts amplifying the frequencies. Then use the **Frequency** knob to locate the optimal frequency where the resonance will most "unpleasant". Then lower the gain as much as needed. Note that in many cases resonances may sound ugly on their own, but they may have an important role in cutting through the mix. Therefore it is advised to use this with caution.

0.00%

Harmonics

Harmonics controls to which extent are the higher harmonics removed along with the main resonance. This is often useful as natural signals are rarely just single frequencies.

0.00%

BW

BW controls the bandwidth of the filter, in other words the range of frequencies the filter affects. Considering the purpose of removing unwanted resonances it is usually best to set it very low, so that it will target the relevant frequencies very specifically.

-40.0 dB

Threshold

Threshold controls the lowest level at which the gate (or more specifically an expander) stops attenuating the sound. Basically, you have 2 options. A) Set the threshold low, which would mean that the gate will only attenuate if the input level is low and won't have any effect on the actual drum sound. In that case you may use high **Ratio** if needed. B) Set the threshold high, which makes the gate become more of an expander processing the overall sound and providing more dynamics. In this case lower **Ratio** is usually needed to get natural results.

2.00 : 1

Ratio

Ratio controls the downwards expander ratio and the higher it is, the more brutal the processing is. Usually the lower the **Threshold** is, the higher the ratio can be for natural results.

300 ms

Release

Release controls how quickly the gate is likely to start attenuating the drum in its sustain stage. In other words, higher values makes the drum sound longer.

GATE

Gate

Gate provides a gating dynamics processor, or more specifically a downwards expander. While it can be used to reduce bleed between the drums, it's alternate purpose is to control the sustain length. For most close mics it should be placed as one of the first processors in the chain, before the dynamics gets changed, and it is used to "stop" the drum sustain at a proper time. In many cases the relatively brutal processing performed by the **Dynamics** and **Maximizer** processors may lead to the length of the drum sound to be unpleasantly long, cluttering the mix and producing an unnatural squashy sound, especially when overused for punchier music styles. The gate may then be used to lower the sustain level before it gets amplified by these processors.

50.0%

Depth

Depth controls the amount of the effect. The higher the value, the more the high frequencies are removed.

500.0 Hz

Frequency

Frequency controls the dynamic filter frequency, above which everything is attenuated unless the drum is hit. The usual setting is around 500Hz, which may seem quite brutal, but usually is surprisingly transparent and produces a solid bleed removal.

100 ms

Release

Release controls the release time of the dynamic filter. The longer it is the longer it takes for the filter to start removing the high frequencies after each hit. Shorter values remove more of the bleed, but potentially also part of the sustain stage of the drum.

BLEEDKILL

BleedKill

BleedKill lets you remove most of the bleed from the cymbals, which are commonly obstructing drum close mics. It employs a dynamic filter, which attenuates everything above selected frequency (mids and highs usually), but once the drum is hit, it temporarily opens and lets the attenuated frequencies pass making the drum sound as it should. This technique is much more subtle than brutal gating you'd probably use otherwise.

Tom bus



Threshold

Threshold controls the lowest level at which the gate (or more specifically an expander) stops attenuating the sound. Basically, you have 2 options. A) Set the threshold low, which would mean that the gate will only attenuate if the input level is low and won't have any effect on the actual drum sound. In that case you may use high **Ratio** if needed. B) Set the threshold high, which makes the gate become more of an expander processing the overall sound and providing more dynamics. In this case lower **Ratio** is usually needed to get natural results.

DYNAMICS

Dynamics

Dynamics controls a powerful dynamics processor, specifically designed for this particular audio material. Unlike normal compressors, which are usually a bit tricky to set up, this one contains multiple units in parallel, each focusing on specific part of the signal. While you can certainly dive into the settings themselves, usually all you need to do is to use the XY pad, which controls the mix ratio of each. For example, when processing a snare drum using the Snare device, 3 units are provided - on focusing on the classic compressor count, another on the initial "pop" used to make the snare cut through the mix, and finally the squasher, which produces very overcompressed mostly sustained sound useful to produce more body. You simply need to set the right ratio between each for your taste.

0.00%

Companding

Companding controls whether the integrated compander is used and how much. Compander is a unit, which compresses the signal when reaching the dynamics section and undoes the compression afterwards. This lets the dynamics section work with more stable levels producing more consistent results. Whether that is good thing or not depends on the music style and actual sound.

100.0%

Depth

Depth controls the amount of the dynamics processing. The plugin provides multiple dynamics processing units focusing on distinct characteristics of the specific drums the device is designer for, e.g. an initial transient + classic compression + squashing. You can then mix these in arbitrary ratio to get the sound you want quickly and easily. The depth then lets you mix this highly processed signal with the dry input. You can see this as parallel compression, though it is far more complex than that.

AUTO
INPUT

Auto-input

Auto-input is the first thing to do with the plugin. Run the audio through the plugin, click the button to enable the analysis, wait a few seconds and disable it. It will automatically adjust the internal input gain to make sure the gain staging in the plugin is ideal. If the device has **Matching** feature, this will also produce the spectrum profile for that to work.



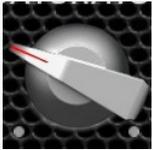
Input

Input controls the input gain and is performed after the **Auto-input** feature. As such this is rarely useful.



Maximizer

Maximizer is a powerful limiter-based maximizer module, which lets you increase loudness of the audio, so that it penetrates the mix better. However please note that for punchy sounds (used in metal for example) the key is to get the initial transients to penetrate the mix and the maximizer may naturally tame these transients as their level is too high compared to the body of the actual drum. So in these cases use this control cautiously and potentially use **Output** control instead. That would make the output exceed 0dBFS, but that shall be fixed later when all the tracks of the drumset are mixed and glued together.



Saturator

Saturator is a powerful analogue style saturation unit, which lets you add some mojo and increase loudness on the way. Be aware that similarly to the **Maximizer** this may tame the wanted transients in more punchy styles, so use this control cautiously.



Output

Output controls the output gain. By default the output will be tuned to reach 0dBFS.



Slap

Slap produces the initial kick transient. It is short, aggressive and potentially high in level. It lets the kick cut through the mix. The more aggressive the music is, the more of this is usually needed.



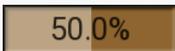
Mix

Mix controls the level of this dynamics processor in the mix.



Drive

Drive controls the aggressivity of the module. Usually the dynamics modules are tuned in such a way that the usual optimum is the default 50%.



Saturation

Saturation controls the amount of the analogue-like saturation of this dynamics processor and that gives the audio some additional "mojo" and can make it more audible in the mix. It can sound heavily distorted when overused however.



Time

Time controls the speed of the dynamics processor. Its effect depends on the specified processor, so the best way to check this is to listen to that specific processor (either by using the XY pad or by disabling the others).



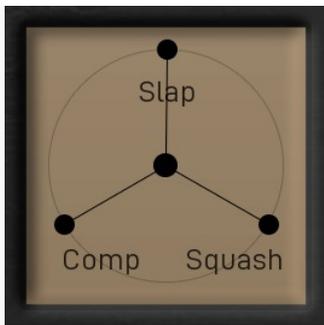
Comp

Comp produces the classic kick compression. It contains the slap, but it mainly makes the kick bigger and longer.



Squash

Squash produces a brutal overcompression, which can be useful to make the kick especially long and big. It is not useful for more aggressive styles as it clutters the mix a lot in most cases.



XY pad

XY pad lets you quickly set the ratio between all the provided dynamics processors.

REVERB

Reverb

Reverb provides and advanced algorithmic reverb chosen specifically for processing this particular type of sound. It may be useful to make the sound wider and bigger. However note that the main effect of spaciousness should come from the room and overhead mics.



Mode

Mode lets you choose from several algorithmic reverb models specifically selected for this type of sound. The modes are usually sorted from the smallest to the biggest and most audible.



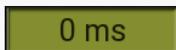
Dry/Wet

Dry/Wet controls the ratio between the dry and reverberated signals. The range is intentionally limited to 40% as that seems a useful maximum for most needs with drums.



Length

Length defines the reverberation signal length. In other words, how long the reverb will become inaudible after the drum hit. In effect this makes the virtual space smaller or bigger.



Predelay

Predelay defines the initial delay before the actual response, which simulates the space between the sound source and the listener. The longer the predelay is, the further away the source seems. At some point (around say 100ms) the brain stops understanding that that the reverberation belongs to the dry signal and starts interpreting them separately, and then the dry signal becomes close to the listener again. Predelay can therefore be used to control the distance from the source to the destination, but detaching the 2 signals can also be useful for instance to fill up a mix that isn't full enough without smearing the signal with the reverb itself.



Low-cut

Low-cut lets you remove the lowest frequencies from the reverb signal. This is often useful to produce cleaner mixes.



High-cut

High-cut lets you remove the highest frequencies from the reverb signal. This is often useful to produce cleaner mixes.



Ratio

Ratio controls the downwards expander ratio and the higher it is, the more brutal the processing is. Usually the lower the **Threshold** is, the higher the ratio can be for natural results.



Release

Release controls how quickly the gate is likely to start attenuating the drum in its sustain stage. In other words, higher values makes the

drum sound longer.

GATE

Gate

Gate provides a gating dynamics processor, or more specifically a downwards expander. While it can be used to reduce bleed between the drums, it's alternate purpose is to control the sustain length. For most close mics it should be placed as one of the first processors in the chain, before the dynamics gets changed, and it is used to "stop" the drum sustain at a proper time. In many cases the relatively brutal processing performed by the **Dynamics** and **Maximizer** processors may lead to the length of the drum sound to be unpleasantly long, cluttering the mix and producing an unnatural squashy sound, especially when overused for punchier music styles. The gate may then be used to lower the sustain level before it gets amplified by these processors.

HiHat



Matching

Matching controls the amount of automatic equalization. It should be performed on a relatively intact signal as it matches your audio to a spectrum of very good recording. Note that in some cases only a little should be applied, such as in case of overheads and rooms, where you only want to ensure there's nothing inherently "different" with your audio.

100.0%

Depth

Depth controls the amount of compression.

-40.0 dB

Threshold

Threshold controls the lowest level at which the gate (or more specifically an expander) stops attenuating the sound. Basically, you have 2 options. A) Set the threshold low, which would mean that the gate will only attenuate if the input level is low and won't have any effect on the actual drum sound. In that case you may use high **Ratio** if needed. B) Set the threshold high, which makes the gate become more of an expander processing the overall sound and providing more dynamics. In this case lower **Ratio** is usually needed to get natural results.

50.0%

Release

Release controls the compressor's release time.

400.0 Hz

Low-cut

Low-cut lets you remove the lower frequencies which usually contain unwanted rumble. For hihat and other cymbals these cuts are usually fairly brutal.

100.0%

Depth

Depth controls the amount of EQ processing. The device provides a specialized equalizer for this specific instrument. If there is a **Fundamental** control, the first step would be to set it up, most likely using the analyzer. That makes the EQ tuned to that sound. The EQ then provides several aspects you can tune the sound with. These are generally specialized filters designed for this specific purpose.

EQ

EQ

EQ provides a powerful dynamic equalizer designed specifically for this particular audio material. Most devices contain the **Fundamental frequency** knob (plus accessible from the integrated analyzer), which you need to set to the lowest and usually most prominent "bump" in the spectrum and which defines the natural resonance frequency of the drum. This makes the bands of the equalizer set to proper locations (which you can further tweak using the **Shift** parameters if needed). Each band then focuses on specific part of the sound and in many cases the defaults will be all that's needed to get a great sound.

COMP

Comp

Comp controls the compressor specially designed for processing hihats. Its purpose isn't really to shape the sound itself, but to make the hits more consistent.

100.0%

Attack

Attack controls the compressor's attack and lowering it will reduce the initial transient, which may or may not be wanted, depending on the music style.

AUTO
INPUT

Auto-input

Auto-input is the first thing to do with the plugin. Run the audio through the plugin, click the button to enable the analysis, wait a few seconds and disable it. It will automatically adjust the internal input gain to make sure the gain staging in the plugin is ideal. If the device has **Matching** feature, this will also produce the spectrum profile for that to work.



Input

Input controls the input gain and is performed after the **Auto-input** feature. As such this is rarely useful.



Maximizer

Maximizer is a powerful limiter-based maximizer module, which lets you increase loudness of the audio, so that it penetrates the mix better. However please note that for punchy sounds (used in metal for example) the key is to get the initial transients to penetrate the mix and the maximizer may naturally tame these transients as their level is too high compared to the body of the actual drum. So in these cases use this control cautiously and potentially use **Output** control instead. That would make the output exceed 0dBFS, but that shall be fixed later when all the tracks of the drumset are mixed and glued together.



Saturator

Saturator is a powerful analogue style saturation unit, which lets you add some mojo and increase loudness on the way. Be aware that similarly to the **Maximizer** this may tame the wanted transients in more punchy styles, so use this control cautiously.



Output

Output controls the output gain. By default the output will be tuned to reach 0dBFS.

BODY

Body

Body controls the middle frequencies usually containing body of the hihat and in most case you will want to cut some of it.



Gain

Gain controls how much the frequencies related to this particular sound characteristic are amplified or reduced, whether statically or dynamically.

50.0%

BW

BW controls the bandwidth of the filter, in other words the range of frequencies the filter affects. If you make it very low, it will target the relevant frequencies very specifically, potentially ringing. If you make it very high, it will change a broader part of the spectrum. The filters are tuned in such a way, that the "usual optimum" is the default 50%.

0.00%

Dynamic

Dynamic controls to which extent is this filter dynamic. When set to 0%, the filter becomes completely static, like if you'd use a traditional parametric equalizer. Increasing this makes the filter move with the sound. 100% makes the filter fully dynamic, meaning that when the input is silent, it has no effect, but once the level exceeds a certain threshold, the filter starts doing its job. This means that the filter will be moving only when the drum is actually hit. This is often very advantageous, since it makes the output much cleaner, not amplifying frequencies longer than actually needed.

Auto

Release

Release controls how quickly the release part of the dynamic EQ ends. The purpose of using a dynamic EQ is to change the sound ONLY during the actual hit. When a drum is hit and the main body filter is engaged, it is usually set to amplify the fundamental frequency. The main body usually ends very quickly though, but the release time can compensate for that by amplifying the frequencies a bit longer, so that the drum sounds bigger. However the release time is too long, the drum may become a bit too long, ringy and obstructing to other elements in the mix.

If the **Dynamic** is set to 0%, this control has no effect.

HARSHNESS

Harhness

Harhness controls the ugly treble frequencies, which are usually removed to make the hihat & cymbals sound cleaner.



Gain

Gain controls how much the frequencies related to this particular sound characteristic are amplified or reduced, whether statically or dynamically.

50.0%

BW

BW controls the bandwidth of the filter, in other words the range of frequencies the filter affects. If you make it very low, it will target the relevant frequencies very specifically, potentially ringing. If you make it very high, it will change a broader part of the spectrum. The filters are tuned in such a way, that the "usual optimum" is the default 50%.

0.00%

Dynamic

Dynamic controls to which extent is this filter dynamic. When set to 0%, the filter becomes completely static, like if you'd use a traditional parametric equalizer. Increasing this makes the filter move with the sound. 100% makes the filter fully dynamic, meaning that when the input is silent, it has no effect, but once the level exceeds a certain threshold, the filter starts doing its job. This means that the filter will be moving only when the drum is actually hit. This is often very advantageous, since it makes the output much cleaner, not amplifying frequencies longer than actually needed.

Auto

Release

Release controls how quickly the release part of the dynamic EQ ends. The purpose of using a dynamic EQ is to change the sound ONLY during the actual hit. When a drum is hit and the main body filter is engaged, it is usually set to amplify the fundamental frequency. The main body usually ends very quickly though, but the release time can compensate for that by amplifying the frequencies a bit longer, so that the drum sounds bigger. However the release time is too long, the drum may become a bit too long, ringy and obstructing to other elements in the mix.

If the **Dynamic** is set to 0%, this control has no effect.

AIR

Air

Air controls the highest frequencies. It may be used to make the hihat more prominent in the mix.



Gain

Gain controls how much the frequencies related to this particular sound characteristic are amplified or reduced, whether statically or dynamically.

50.0%

BW

BW controls the bandwidth of the filter, in other words the range of frequencies the filter affects. If you make it very low, it will target the relevant frequencies very specifically, potentially ringing. If you make it very high, it will change a broader part of the spectrum. The filters are tuned in such a way, that the "usual optimum" is the default 50%.

0.00%

Dynamic

Dynamic controls to which extent is this filter dynamic. When set to 0%, the filter becomes completely static, like if you'd use a traditional parametric equalizer. Increasing this makes the filter move with the sound. 100% makes the filter fully dynamic, meaning that when the input is silent, it has no effect, but once the level exceeds a certain threshold, the filter starts doing its job. This means that the filter will be moving only when the drum is actually hit. This is often very advantageous, since it makes the output much cleaner, not amplifying frequencies longer than actually needed.

Auto

Release

Release controls how quickly the release part of the dynamic EQ ends. The purpose of using a dynamic EQ is to change the sound ONLY during the actual hit. When a drum is hit and the main body filter is engaged, it is usually set to amplify the fundamental frequency. The main body usually ends very quickly though, but the release time can compensate for that by amplifying the frequencies a bit longer, so that the drum sounds bigger. However the release time is too long, the drum may become a bit too long, ringy and obstructing to other elements in the mix.

If the **Dynamic** is set to 0%, this control has no effect.

RESOKILL

Resokill

Resokill is a resonance removal module based on 2 very narrow peak filters, which let you remove unwanted resonances many drums suffer from. The knobs control the frequencies of these filters and you can also see them as red lines in the analyzer graph.



Frequency

Frequency controls the resonance frequency to be removed. You can see the location as the red line in the analyzer graph. Resonances often need to be set very accurately, so that the filter is most efficient and natural, so don't forget to use mouse-wheel or ctrl/cmd for fine-tuning.

0.00 dB

Gain

Gain controls the gain of the resonance filter. The usual workflow is to set this to maximum, so that the filter starts amplifying the frequencies. Then use the **Frequency** knob to locate the optimal frequency where the resonance will most "unpleasant". Then lower the gain as much as needed. Note that in many cases resonances may sound ugly on their own, but they may have an important role in cutting through the mix. Therefore it is advised to use this with caution.

0.00%

Harmonics

Harmonics controls to which extent are the higher harmonics removed along with the main resonance. This is often useful as natural signals are rarely just single frequencies.

0.00%

BW

BW controls the bandwidth of the filter, in other words the range of frequencies the filter affects. Considering the purpose of removing unwanted resonances it is usually best to set it very low, so that it will target the relevant frequencies very specifically.

2.00 : 1

Ratio

Ratio controls the downwards expander ratio and the higher it is, the more brutal the processing is. Usually the lower the **Threshold** is, the higher the ratio can be for natural results.

400 ms

Release

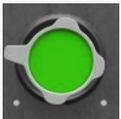
Release controls how quickly the gate is likely to start attenuating the drum in its sustain stage. In other words, higher values makes the drum sound longer.

GATE

Gate

Gate provides a gating dynamics processor, or more specifically a downwards expander. While it can be used to reduce bleed between the drums, it's alternate purpose is to control the sustain length. For most close mics it should be placed as one of the first processors in the chain, before the dynamics gets changed, and it is used to "stop" the drum sustain at a proper time. In many cases the relatively brutal processing performed by the **Dynamics** and **Maximizer** processors may lead to the length of the drum sound to be unpleasantly long, cluttering the mix and producing an unnatural squashy sound, especially when overused for punchier music styles. The gate may then be used to lower the sustain level before it gets amplified by these processors.

Overheads



Matching

Matching controls the amount of automatic equalization. It should be performed on a relatively intact signal as it matches your audio to a spectrum of very good recording. Note that in some cases only a little should be applied, such as in case of overheads and rooms, where you only want to ensure there's nothing inherently "different" with your audio.



Dark / Bright

Dark / Bright controls the overall brightness - ratio of higher frequencies compared to lower frequencies. It is often useful to increase this for room mics, which are sometimes a bit dull due to air absorption.

Mode



Mode lets you choose from several algorithmic reverb models specifically selected for this type of sound. The modes are usually sorted from the smallest to the biggest and most audible.



Low-cut

Low-cut lets you remove the low frequencies from the overhead / room tracks. Usually this is required to keep the bass spectrum clean from any ambience.



Depth

Depth controls the amount of EQ processing. The device provides a specialized equalizer for this specific instrument. If there is a **Fundamental** control, the first step would be to set it up, most likely using the analyzer. That makes the EQ tuned to that sound. The EQ then provides several aspects you can tune the sound with. These are generally specialized filters designed for this specific purpose.



EQ

EQ provides a powerful dynamic equalizer designed specifically for this particular audio material. Most devices contain the **Fundamental frequency** knob (plus accessible from the integrated analyzer), which you need to set to the lowest and usually most prominent "bump" in the spectrum and which defines the natural resonance frequency of the drum. This makes the bands of the equalizer set to proper locations (which you can further tweak using the **Shift** parameters if needed). Each band then focuses on specific part of the sound and in many cases the defaults will be all that's needed to get a great sound.



Dynamics

Dynamics controls a powerful dynamics processor, specifically designed for this particular audio material. Unlike normal compressors, which are usually a bit tricky to set up, this one contains multiple units in parallel, each focusing on specific part of the signal. While you can certainly dive into the settings themselves, usually all you need to do is to use the XY pad, which controls the mix ratio of each. For example, when processing a snare drum using the Snare device, 3 units are provided - one focusing on the classic compressor count, another on the initial "pop" used to make the snare cut through the mix, and finally the squasher, which produces very overcompressed mostly sustained sound useful to produce more body. You simply need to set the right ratio between each for your taste.

0.00%

Companing

Companing controls whether the integrated compander is used and how much. Compander is a unit, which compresses the signal when reaching the dynamics section and undoes the compression afterwards. This lets the dynamics section work with more stable levels producing more consistent results. Whether that is good thing or not depends on the music style and actual sound.

100.0%

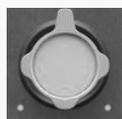
Depth

Depth controls the amount of the dynamics processing. The plugin provides multiple dynamics processing units focusing on distinct characteristics of the specific drums the device is designer for, e.g. an initial transient + classic compression + squashing. You can then mix these in arbitrary ratio to get the sound you want quickly and easily. The depth then lets you mix this highly processed signal with the dry input. You can see this as parallel compression, though it is far more complex than that.

AUTO
INPUT

Auto-input

Auto-input is the first thing to do with the plugin. Run the audio through the plugin, click the button to enable the analysis, wait a few seconds and disable it. It will automatically adjust the internal input gain to make sure the gain staging in the plugin is ideal. If the device has **Matching** feature, this will also produce the spectrum profile for that to work.



Input

Input controls the input gain and is performed after the **Auto-input** feature. As such this is rarely useful.



Maximizer

Maximizer is a powerful limiter-based maximizer module, which lets you increase loudness of the audio, so that it penetrates the mix better. However please note that for punchy sounds (used in metal for example) the key is to get the initial transients to penetrate the mix and the maximizer may naturally tame these transients as their level is too high compared to the body of the actual drum. So in these cases use this control cautiously and potentially use **Output** control instead. That would make the output exceed 0dBFS, but that shall be fixed later when all the tracks of the drumset are mixed and glued together.



Saturator

Saturator is a powerful analogue style saturation unit, which lets you add some mojo and increase loudness on the way. Be aware that similarly to the **Maximizer** this may tame the wanted transients in more punchy styles, so use this control cautiously.



Output

Output controls the output gain. By default the output will be tuned to reach 0dBFS.

KICK

Kick

Kick controls the bass part of the spectrum, where mainly kick & toms are located.



Gain

Gain controls how much the frequencies related to this particular sound characteristic are amplified or reduced, whether statically or dynamically.

50.0%

BW

BW controls the bandwidth of the filter, in other words the range of frequencies the filter affects. If you make it very low, it will target the relevant frequencies very specifically, potentially ringing. If you make it very high, it will change a broader part of the spectrum. The filters are tuned in such a way, that the "usual optimum" is the default 50%.

Off

Dynamic

Dynamic controls to which extent is this filter dynamic. When set to 0%, the filter becomes completely static, like if you'd use a

traditional parametric equalizer. Increasing this makes the filter move with the sound. 100% makes the filter fully dynamic, meaning that when the input is silent, it has no effect, but once the level exceeds a certain threshold, the filter starts doing its job. This means that the filter will be moving only when the drum is actually hit. This is often very advantageous, since it makes the output much cleaner, not amplifying frequencies longer than actually needed.

Auto

Release

Release controls how quickly the release part of the dynamic EQ ends. The purpose of using a dynamic EQ is to change the sound ONLY during the actual hit. When a drum is hit and the main body filter is engaged, it is usually set to amplify the fundamental frequency. The main body usually ends very quickly though, but the release time can compensate for that by amplifying the frequencies a bit longer, so that the drum sounds bigger. However the release time is too long, the drum may become a bit too long, ringy and obstructing to other elements in the mix.

If the **Dynamic** is set to 0%, this control has no effect.

BASS

Bass

Bass controls the low frequency part of the spectrum, where the snare and toms are mainly located.



Gain

Gain controls how much the frequencies related to this particular sound characteristic are amplified or reduced, whether statically or dynamically.

50.0%

BW

BW controls the bandwidth of the filter, in other words the range of frequencies the filter affects. If you make it very low, it will target the relevant frequencies very specifically, potentially ringing. If you make it very high, it will change a broader part of the spectrum. The filters are tuned in such a way, that the "usual optimum" is the default 50%.

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Dynamic

Dynamic controls to which extent is this filter dynamic. When set to 0%, the filter becomes completely static, like if you'd use a traditional parametric equalizer. Increasing this makes the filter move with the sound. 100% makes the filter fully dynamic, meaning that when the input is silent, it has no effect, but once the level exceeds a certain threshold, the filter starts doing its job. This means that the filter will be moving only when the drum is actually hit. This is often very advantageous, since it makes the output much cleaner, not amplifying frequencies longer than actually needed.

Auto

Release

Release controls how quickly the release part of the dynamic EQ ends. The purpose of using a dynamic EQ is to change the sound ONLY during the actual hit. When a drum is hit and the main body filter is engaged, it is usually set to amplify the fundamental frequency. The main body usually ends very quickly though, but the release time can compensate for that by amplifying the frequencies a bit longer, so that the drum sounds bigger. However the release time is too long, the drum may become a bit too long, ringy and obstructing to other elements in the mix.

If the **Dynamic** is set to 0%, this control has no effect.

MIDS

Mids

Mids controls the high mid part of the spectrum, where brighter part of the snare is mainly located.



Gain

Gain controls how much the frequencies related to this particular sound characteristic are amplified or reduced, whether statically or dynamically.

50.0%

BW

BW controls the bandwidth of the filter, in other words the range of frequencies the filter affects. If you make it very low, it will target the relevant frequencies very specifically, potentially ringing. If you make it very high, it will change a broader part of the spectrum. The filters are tuned in such a way, that the "usual optimum" is the default 50%.

Off

Dynamic

Dynamic controls to which extent is this filter dynamic. When set to 0%, the filter becomes completely static, like if you'd use a traditional parametric equalizer. Increasing this makes the filter move with the sound. 100% makes the filter fully dynamic, meaning that when the input is silent, it has no effect, but once the level exceeds a certain threshold, the filter starts doing its job. This means that the filter will be moving only when the drum is actually hit. This is often very advantageous, since it makes the output much cleaner, not amplifying frequencies longer than actually needed.

Auto

Release

Release controls how quickly the release part of the dynamic EQ ends. The purpose of using a dynamic EQ is to change the sound ONLY during the actual hit. When a drum is hit and the main body filter is engaged, it is usually set to amplify the fundamental frequency. The main body usually ends very quickly though, but the release time can compensate for that by amplifying the frequencies a bit longer, so that the drum sounds bigger. However the release time is too long, the drum may become a bit too long, ringy and obstructing to other

elements in the mix.
If the **Dynamic** is set to 0%, this control has no effect.

TREBLE Treble

Treble controls the high frequencies.



Gain

Gain controls how much the frequencies related to this particular sound characteristic are amplified or reduced, whether statically or dynamically.

50.0% BW

BW controls the bandwidth of the filter, in other words the range of frequencies the filter affects. If you make it very low, it will target the relevant frequencies very specifically, potentially ringing. If you make it very high, it will change a broader part of the spectrum. The filters are tuned in such a way, that the "usual optimum" is the default 50%.

Off Dynamic

Dynamic controls to which extent is this filter dynamic. When set to 0%, the filter becomes completely static, like if you'd use a traditional parametric equalizer. Increasing this makes the filter move with the sound. 100% makes the filter fully dynamic, meaning that when the input is silent, it has no effect, but once the level exceeds a certain threshold, the filter starts doing its job. This means that the filter will be moving only when the drum is actually hit. This is often very advantageous, since it makes the output much cleaner, not amplifying frequencies longer than actually needed.

Auto Release

Release controls how quickly the release part of the dynamic EQ ends. The purpose of using a dynamic EQ is to change the sound ONLY during the actual hit. When a drum is hit and the main body filter is engaged, it is usually set to amplify the fundamental frequency. The main body usually ends very quickly though, but the release time can compensate for that by amplifying the frequencies a bit longer, so that the drum sounds bigger. However the release time is too long, the drum may become a bit too long, ringy and obstructing to other elements in the mix.

If the **Dynamic** is set to 0%, this control has no effect.

AIR Air

Air controls the highest frequencies. It may be tempting to increase this range to make the drums sound better on their own, but note that in many cases that would clutter that part of the mix too much.



Gain

Gain controls how much the frequencies related to this particular sound characteristic are amplified or reduced, whether statically or dynamically.

50.0% BW

BW controls the bandwidth of the filter, in other words the range of frequencies the filter affects. If you make it very low, it will target the relevant frequencies very specifically, potentially ringing. If you make it very high, it will change a broader part of the spectrum. The filters are tuned in such a way, that the "usual optimum" is the default 50%.

Off Dynamic

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Auto Release

Release controls how quickly the release part of the dynamic EQ ends. The purpose of using a dynamic EQ is to change the sound ONLY during the actual hit. When a drum is hit and the main body filter is engaged, it is usually set to amplify the fundamental frequency. The main body usually ends very quickly though, but the release time can compensate for that by amplifying the frequencies a bit longer, so that the drum sounds bigger. However the release time is too long, the drum may become a bit too long, ringy and obstructing to other elements in the mix.

If the **Dynamic** is set to 0%, this control has no effect.

RESOKILL Resokill

Resokill is a resonance removal module based on 2 very narrow peak filters, which let you remove unwanted resonances many drums suffer from. The knobs control the frequencies of these filters and you can also see them as red lines in the analyzer graph.



Frequency

Frequency controls the resonance frequency to be removed. You can see the location as the red line in the analyzer graph. Resonances often need to be set very accurately, so that the filter is most efficient and natural, so don't forget to use mouse-wheel or ctrl/cmd for fine-tuning.

0.00 dB

Gain

Gain controls the gain of the resonance filter. The usual workflow is to set this to maximum, so that the filter starts amplifying the frequencies. Then use the **Frequency** knob to locate the optimal frequency where the resonance will most "unpleasant". Then lower the gain as much as needed. Note that in many cases resonances may sound ugly on their own, but they may have an important role in cutting through the mix. Therefore it is advised to use this with caution.

0.00%

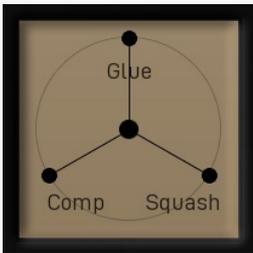
Harmonics

Harmonics controls to which extent are the higher harmonics removed along with the main resonance. This is often useful as natural signals are rarely just single frequencies.

0.00%

BW

BW controls the bandwidth of the filter, in other words the range of frequencies the filter affects. Considering the purpose of removing unwanted resonances it is usually best to set it very low, so that it will target the relevant frequencies very specifically.



XY pad

XY pad lets you quickly set the ratio between all the provided dynamics processors.

GLUE

Pop

Pop produces the initial snare peak. It is short, aggressive and potentially high in level. It lets the snare drum cut through the mix. The more aggressive the music is, the more of this is usually needed.



Mix

Mix controls the level of this dynamics processor in the mix.



Drive

Drive controls the aggressivity of the module. Usually the dynamics modules are tuned in such a way that the usual optimum is the default 50%.

20.0%

Saturation

Saturation controls the amount of the analogue-like saturation of this dynamics processor and that gives the audio some additional "mojo" and can make it more audible in the mix. It can sound heavily distorted when overused however.

50.0%

Time

Time controls the speed of the dynamics processor. Its effect depends on the specified processor, so the best way to check this is to listen to that specific processor (either by using the XY pad or by disabling the others).

COMP

Comp

Comp produces the classic snare compression. It contains the pop, but it mainly makes the snare sound bigger and longer.

SQUASH

Squash

Squash produces a brutal overcompression, which can be useful to make the snare especially long and big. It is not useful for more aggressive styles as it clutters the mix a lot in most cases.

REVERB

Reverb

Reverb provides an advanced algorithmic reverb chosen specifically for processing this particular type of sound. It may be useful to make the sound wider and bigger. However note that the main effect of spaciousness should come from the room and overhead mics.



Dry/Wet

Dry/Wet controls the ratio between the dry and reverberated signals. The range is intentionally limited to 40% as that seems a useful maximum for most needs with drums.



Length

Length defines the reverberation signal length. In other words, how long the reverb will become inaudible after the drum hit. In effect this makes the virtual space smaller or bigger.

0 ms

Predelay

Predelay defines the initial delay before the actual response, which simulates the space between the sound source and the listener. The longer the predelay is, the further away the source seems. At some point (around say 100ms) the brain stops understanding that that the reverberation belongs to the dry signal and starts interpreting them separately, and then the dry signal becomes close to the listener again. Predelay can therefore be used to control the distance from the source to the destination, but detaching the 2 signals can also be useful for instance to fill up a mix that isn't full enough without smearing the signal with the reverb itself.

200.0 Hz

Low-cut

Low-cut lets you remove the lowest frequencies from the reverb signal. This is often useful to produce cleaner mixes.

Off

High-cut

High-cut lets you remove the highest frequencies from the reverb signal. This is often useful to produce cleaner mixes.

50.0%

Depth

Depth controls the amount of the dynamic filtering.

500.0 Hz

Frequency

Frequency controls the filter frequency above which things are attenuated unless low end is present.

150 ms

Release

Release controls how quickly the high end gets attenuated again after a drum hit. Lower values provide remove the cymbals more, but may change the drum sound as well.

CKILL

CymbalKill

CymbalKill lets you dynamically attenuate the cymbals by using a dynamic filter that removes high end unless low end (kick / snare...) is present. This lets you get more drum signal out of the overheads and rooms, which would otherwise be drowning in cymbals.

Room

The screenshot shows a detailed digital audio workstation interface for a 'Room' reverb plugin. The interface is organized into several functional sections:

- Left Panel:** Contains global controls including 'AUTO INPUT', 'INPUT', 'MATCHING', 'ROOM' (brand name), 'MAXIMIZER', and 'SATURATOR'.
- EQ Section:** Features a frequency response graph with a peak at 1418 Hz. Below the graph are five frequency sliders labeled 'KICK', 'BASS', 'MIDS', 'TREBLE', and 'AIR'. A 'RESOKILL' section with two gain knobs is also present.
- Dynamics Section:** Includes a 'DEPTH' knob set to 100.0%, a 'COMPANDING' knob set to 0.00%, and three knobs for 'GLUE', 'COMP', and 'SQUASH'. A 'SATURATION' table is shown below with columns for '20.0%', '20.0%', and '20.0%' and a 'TIME' row set to '50.0%', '50.0%', and '50.0%'. A small diagram illustrates the relationship between 'Glue', 'Comp', and 'Squash'.
- Reverb Section:** Features a 'CHAMBER' slider, 'DRY/WET' and 'LENGTH' knobs, and three buttons for 'PREDELAY' (0 ms), 'LOW-CUT' (200.0 Hz), and 'HIGH-CUT' (Off).
- CKILL Section:** Includes 'DEPTH' (50.0%), 'FREQUENCY' (500.0 Hz), and 'RELEASE' (150 ms) controls.



Matching

Matching controls the amount of automatic equalization. It should be performed on a relatively intact signal as it matches your audio to a spectrum of very good recording. Note that in some cases only a little should be applied, such as in case of overheads and rooms, where you only want to ensure there's nothing inherently "different" with your audio.

0.00 dB

Dark / Bright

Dark / Bright controls the overall brightness - ratio of higher frequencies compared to lower frequencies. It is often useful to increase this for room mics, which are sometimes a bit dull due to air absorption.

Mode



Mode lets you choose from several algorithmic reverb models specifically selected for this type of sound. The modes are usually sorted from the smallest to the biggest and most audible.

100.00 Hz

Low-cut

Low-cut lets you remove the low frequencies from the overhead / room tracks. Usually this is required to keep the bass spectrum clean from any ambience.

100.0%

Depth

Depth controls the amount of EQ processing. The device provides a specialized equalizer for this specific instrument. If there is a **Fundamental** control, the first step would be to set it up, most likely using the analyzer. That makes the EQ tuned to that sound. The EQ then provides several aspects you can tune the sound with. These are generally specialized filters designed for this specific purpose.

EQ

EQ

EQ provides a powerful dynamic equalizer designed specifically for this particular audio material. Most devices contain the **Fundamental frequency** knob (plus accessible from the integrated analyzer), which you need to set to the lowest and usually most prominent "bump" in the spectrum and which defines the natural resonance frequency of the drum. This makes the bands of the equalizer set to proper locations (which you can further tweak using the **Shift** parameters if needed). Each band then focuses on specific part of the sound and in many cases the defaults will be all that's needed to get a great sound.

DYNAMICS

Dynamics

Dynamics controls a powerful dynamics processor, specifically designed for this particular audio material. Unlike normal compressors, which are usually a bit tricky to set up, this one contains multiple units in parallel, each focusing on specific part of the signal. While you can certainly dive into the settings themselves, usually all you need to do is to use the XY pad, which controls the mix ratio of each. For example, when processing a snare drum using the Snare device, 3 units are provided - on focusing on the classic compressor count, another on the initial "pop" used to make the snare cut through the mix, and finally the squasher, which produces very overcompressed mostly sustained sound useful to produce more body. You simply need to set the right ratio between each for your taste.

0.00%

Companing

Companing controls whether the integrated compander is used and how much. Compander is a unit, which compresses the signal when reaching the dynamics section and undoes the compression afterwards. This lets the dynamics section work with more stable levels producing more consistent results. Whether that is good thing or not depends on the music style and actual sound.

100.0%

Depth

Depth controls the amount of the dynamics processing. The plugin provides multiple dynamics processing units focusing on distinct characteristics of the specific drums the device is designer for, e.g. an initial transient + classic compression + squashing. You can then mix these in arbitrary ratio to get the sound you want quickly and easily. The depth then lets you mix this highly processed signal with the dry input. You can see this as parallel compression, though it is far more complex than that.

AUTO
INPUT

Auto-input

Auto-input is the first thing to do with the plugin. Run the audio through the plugin, click the button to enable the analysis, wait a few seconds and disable it. It will automatically adjust the internal input gain to make sure the gain staging in the plugin is ideal. If the device has **Matching** feature, this will also produce the spectrum profile for that to work.



Input

Input controls the input gain and is performed after the **Auto-input** feature. As such this is rarely useful.



Maximizer

Maximizer is a powerful limiter-based maximizer module, which lets you increase loudness of the audio, so that it penetrates the mix better. However please note that for punchy sounds (used in metal for example) the key is to get the initial transients to penetrate the mix and the maximizer may naturally tame these transients as their level is too high compared to the body of the actual drum. So in these cases use this control cautiously and potentially use **Output** control instead. That would make the output exceed 0dBFS, but that shall be fixed later when all the tracks of the drumset are mixed and glued together.



Saturator

Saturator is a powerful analogue style saturation unit, which lets you add some mojo and increase loudness on the way. Be aware that similarly to the **Maximizer** this may tame the wanted transients in more punchy styles, so use this control cautiously.



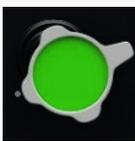
Output

Output controls the output gain. By default the output will be tuned to reach 0dBFS.



Kick

Kick controls the bass part of the spectrum, where mainly kick & toms are located.



Gain

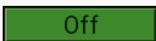
Gain controls how much the frequencies related to this particular sound characteristic are amplified or reduced, whether statically or dynamically.



50.0%

BW

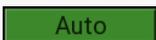
BW controls the bandwidth of the filter, in other words the range of frequencies the filter affects. If you make it very low, it will target the relevant frequencies very specifically, potentially ringing. If you make it very high, it will change a broader part of the spectrum. The filters are tuned in such a way, that the "usual optimum" is the default 50%.



Off

Dynamic

Dynamic controls to which extent is this filter dynamic. When set to 0%, the filter becomes completely static, like if you'd use a traditional parametric equalizer. Increasing this makes the filter move with the sound. 100% makes the filter fully dynamic, meaning that when the input is silent, it has no effect, but once the level exceeds a certain threshold, the filter starts doing its job. This means that the filter will be moving only when the drum is actually hit. This is often very advantageous, since it makes the output much cleaner, not amplifying frequencies longer than actually needed.

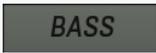


Auto

Release

Release controls how quickly the release part of the dynamic EQ ends. The purpose of using a dynamic EQ is to change the sound **ONLY** during the actual hit. When a drum is hit and the main body filter is engaged, it is usually set to amplify the fundamental frequency. The main body usually ends very quickly though, but the release time can compensate for that by amplifying the frequencies a bit longer, so that the drum sounds bigger. However the release time is too long, the drum may become a bit too long, ringy and obstructing to other elements in the mix.

If the **Dynamic** is set to 0%, this control has no effect.



BASS

Bass

Bass controls the low frequency part of the spectrum, where the snare and toms are mainly located.



Gain

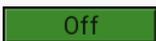
Gain controls how much the frequencies related to this particular sound characteristic are amplified or reduced, whether statically or dynamically.



50.0%

BW

BW controls the bandwidth of the filter, in other words the range of frequencies the filter affects. If you make it very low, it will target the relevant frequencies very specifically, potentially ringing. If you make it very high, it will change a broader part of the spectrum. The filters are tuned in such a way, that the "usual optimum" is the default 50%.



Off

Dynamic

Dynamic controls to which extent is this filter dynamic. When set to 0%, the filter becomes completely static, like if you'd use a

traditional parametric equalizer. Increasing this makes the filter move with the sound. 100% makes the filter fully dynamic, meaning that when the input is silent, it has no effect, but once the level exceeds a certain threshold, the filter starts doing its job. This means that the filter will be moving only when the drum is actually hit. This is often very advantageous, since it makes the output much cleaner, not amplifying frequencies longer than actually needed.

Auto

Release

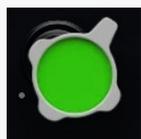
Release controls how quickly the release part of the dynamic EQ ends. The purpose of using a dynamic EQ is to change the sound ONLY during the actual hit. When a drum is hit and the main body filter is engaged, it is usually set to amplify the fundamental frequency. The main body usually ends very quickly though, but the release time can compensate for that by amplifying the frequencies a bit longer, so that the drum sounds bigger. However the release time is too long, the drum may become a bit too long, ringy and obstructing to other elements in the mix.

If the **Dynamic** is set to 0%, this control has no effect.

MIDS

Mids

Mids controls the high mid part of the spectrum, where brighter part of the snare is mainly located.



Gain

Gain controls how much the frequencies related to this particular sound characteristic are amplified or reduced, whether statically or dynamically.

50.0%

BW

BW controls the bandwidth of the filter, in other words the range of frequencies the filter affects. If you make it very low, it will target the relevant frequencies very specifically, potentially ringing. If you make it very high, it will change a broader part of the spectrum. The filters are tuned in such a way, that the "usual optimum" is the default 50%.

Off

Dynamic

Dynamic controls to which extent is this filter dynamic. When set to 0%, the filter becomes completely static, like if you'd use a traditional parametric equalizer. Increasing this makes the filter move with the sound. 100% makes the filter fully dynamic, meaning that when the input is silent, it has no effect, but once the level exceeds a certain threshold, the filter starts doing its job. This means that the filter will be moving only when the drum is actually hit. This is often very advantageous, since it makes the output much cleaner, not amplifying frequencies longer than actually needed.

Auto

Release

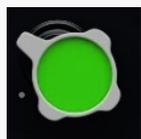
Release controls how quickly the release part of the dynamic EQ ends. The purpose of using a dynamic EQ is to change the sound ONLY during the actual hit. When a drum is hit and the main body filter is engaged, it is usually set to amplify the fundamental frequency. The main body usually ends very quickly though, but the release time can compensate for that by amplifying the frequencies a bit longer, so that the drum sounds bigger. However the release time is too long, the drum may become a bit too long, ringy and obstructing to other elements in the mix.

If the **Dynamic** is set to 0%, this control has no effect.

TREBLE

Treble

Treble controls the high frequencies.



Gain

Gain controls how much the frequencies related to this particular sound characteristic are amplified or reduced, whether statically or dynamically.

50.0%

BW

BW controls the bandwidth of the filter, in other words the range of frequencies the filter affects. If you make it very low, it will target the relevant frequencies very specifically, potentially ringing. If you make it very high, it will change a broader part of the spectrum. The filters are tuned in such a way, that the "usual optimum" is the default 50%.

Off

Dynamic

Dynamic controls to which extent is this filter dynamic. When set to 0%, the filter becomes completely static, like if you'd use a traditional parametric equalizer. Increasing this makes the filter move with the sound. 100% makes the filter fully dynamic, meaning that when the input is silent, it has no effect, but once the level exceeds a certain threshold, the filter starts doing its job. This means that the filter will be moving only when the drum is actually hit. This is often very advantageous, since it makes the output much cleaner, not amplifying frequencies longer than actually needed.

Auto

Release

Release controls how quickly the release part of the dynamic EQ ends. The purpose of using a dynamic EQ is to change the sound ONLY during the actual hit. When a drum is hit and the main body filter is engaged, it is usually set to amplify the fundamental frequency. The main body usually ends very quickly though, but the release time can compensate for that by amplifying the frequencies a bit longer, so that the drum sounds bigger. However the release time is too long, the drum may become a bit too long, ringy and obstructing to other

elements in the mix.

If the **Dynamic** is set to 0%, this control has no effect.

AIR

Air

Air controls the highest frequencies. It may be tempting to increase this range to make the drums sound better on their own, but note that in many cases that would clutter that part of the mix too much.



Gain

Gain controls how much the frequencies related to this particular sound characteristic are amplified or reduced, whether statically or dynamically.

50.0%

BW

BW controls the bandwidth of the filter, in other words the range of frequencies the filter affects. If you make it very low, it will target the relevant frequencies very specifically, potentially ringing. If you make it very high, it will change a broader part of the spectrum. The filters are tuned in such a way, that the "usual optimum" is the default 50%.

Off

Dynamic

Dynamic controls to which extent is this filter dynamic. When set to 0%, the filter becomes completely static, like if you'd use a traditional parametric equalizer. Increasing this makes the filter move with the sound. 100% makes the filter fully dynamic, meaning that when the input is silent, it has no effect, but once the level exceeds a certain threshold, the filter starts doing its job. This means that the filter will be moving only when the drum is actually hit. This is often very advantageous, since it makes the output much cleaner, not amplifying frequencies longer than actually needed.

Auto

Release

Release controls how quickly the release part of the dynamic EQ ends. The purpose of using a dynamic EQ is to change the sound **ONLY** during the actual hit. When a drum is hit and the main body filter is engaged, it is usually set to amplify the fundamental frequency. The main body usually ends very quickly though, but the release time can compensate for that by amplifying the frequencies a bit longer, so that the drum sounds bigger. However the release time is too long, the drum may become a bit too long, ringy and obstructing to other elements in the mix.

If the **Dynamic** is set to 0%, this control has no effect.

RESOKILL

Resokill

Resokill is a resonance removal module based on 2 very narrow peak filters, which let you remove unwanted resonances many drums suffer from. The knobs control the frequencies of these filters and you can also see them as red lines in the analyzer graph.



Frequency

Frequency controls the resonance frequency to be removed. You can see the location as the red line in the analyzer graph. Resonances often need to be set very accurately, so that the filter is most efficient and natural, so don't forget to use mouse-wheel or ctrl/cmd for fine-tuning.

0.00 dB

Gain

Gain controls the gain of the resonance filter. The usual workflow is to set this to maximum, so that the filter starts amplifying the frequencies. Then use the **Frequency** knob to locate the optimal frequency where the resonance will most "unpleasant". Then lower the gain as much as needed. Note that in many cases resonances may sound ugly on their own, but they may have an important role in cutting through the mix. Therefore it is advised to use this with caution.

0.00%

Harmonics

Harmonics controls to which extent are the higher harmonics removed along with the main resonance. This is often useful as natural signals are rarely just single frequencies.

0.00%

BW

BW controls the bandwidth of the filter, in other words the range of frequencies the filter affects. Considering the purpose of removing unwanted resonances it is usually best to set it very low, so that it will target the relevant frequencies very specifically.



XY pad

XY pad lets you quickly set the ratio between all the provided dynamics processors.

GLUE

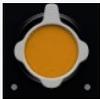
Pop

Pop produces the initial snare peak. It is short, aggressive and potentially high in level. It lets the snare drum cut through the mix. The more aggressive the music is, the more of this is usually needed.



Mix

Mix controls the level of this dynamics processor in the mix.



Drive

Drive controls the aggressivity of the module. Usually the dynamics modules are tuned in such a way that the usual optimum is the default 50%.

20.0%

Saturation

Saturation controls the amount of the analogue-like saturation of this dynamics processor and that gives the audio some additional "mojo" and can make it more audible in the mix. It can sound heavily distorted when overused however.

50.0%

Time

Time controls the speed of the dynamics processor. Its effect depends on the specified processor, so the best way to check this is to listen to that specific processor (either by using the XY pad or by disabling the others).

COMP

Comp

Comp produces the classic snare compression. It contains the pop, but it mainly makes the snare sound bigger and longer.

SQUASH

Squash

Squash produces a brutal overcompression, which can be useful to make the snare especially long and big. It is not useful for more aggressive styles as it clutters the mix a lot in most cases.

REVERB

Reverb

Reverb provides an advanced algorithmic reverb chosen specifically for processing this particular type of sound. It may be useful to make the sound wider and bigger. However note that the main effect of spaciousness should come from the room and overhead mics.



Dry/Wet

Dry/Wet controls the ratio between the dry and reverberated signals. The range is intentionally limited to 40% as that seems a useful maximum for most needs with drums.



Length

Length defines the reverberation signal length. In other words, how long the reverb will become inaudible after the drum hit. In effect this makes the virtual space smaller or bigger.

0 ms

Predelay

Predelay defines the initial delay before the actual response, which simulates the space between the sound source and the listener. The longer the predelay is, the further away the source seems. At some point (around say 100ms) the brain stops understanding that the reverberation belongs to the dry signal and starts interpreting them separately, and then the dry signal becomes close to the listener again. Predelay can therefore be used to control the distance from the source to the destination, but detaching the 2 signals can also be useful for instance to fill up a mix that isn't full enough without smearing the signal with the reverb itself.

200.0 Hz

Low-cut

Low-cut lets you remove the lowest frequencies from the reverb signal. This is often useful to produce cleaner mixes.

Off

High-cut

High-cut lets you remove the highest frequencies from the reverb signal. This is often useful to produce cleaner mixes.

50.0%

Depth

Depth controls the amount of the dynamic filtering.

500.0 Hz

Frequency

Frequency controls the filter frequency above which things are attenuated unless low end is present.

150 ms

Release

Release controls how quickly the high end gets attenuated again after a drum hit. Lower values provide remove the cymbals more, but may change the drum sound as well.

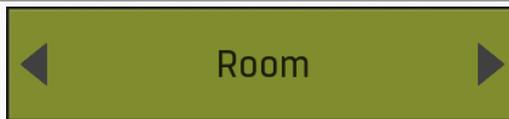
CKILL CymbalKill

CymbalKill lets you dynamically attenuate the cymbals by using a dynamic filter that removes high end unless low end (kick / snare...) is present. This lets you get more drum signal out of the overheads and rooms, which would otherwise be drowning in cymbals.

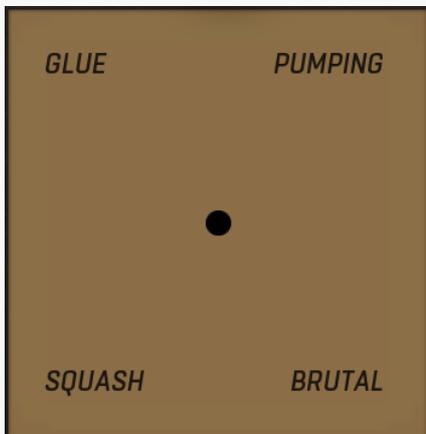
Parallel



Mode



Mode provides several different reverb modes sorted from the smallest to the largest. Use a bigger reverb if there is more "space to fill".



XY

XY pad lets you control ratios of all compressors at once, which makes searching for the right "character" very fast.



Brutal

Brutal controls the amount of the "brutal" compression based on specific a analog compressor simulation with this specific type of sound.



Squash

Squash controls the amount of the squashing compression based on a specific analog compressor simulation with this specific type of sound.



Pumping

Pumping controls the amount of the pumping compression based on a specific analog compressor simulation with this specific type of sound.



Bass

Bass lets you amplify or attenuate the lower frequencies of the reverb signal.



Treble

Treble lets you amplify or attenuate the high frequencies of the reverb signal.



Auto-input

Auto-input is the first thing to do with the plugin. Run the audio through the plugin, click the button to enable the analysis, wait a few seconds and disable it. It will automatically adjust the internal input gain to make sure the gain staging in the plugin is ideal. If the device has **Matching** feature, this will also produce the spectrum profile for that to work.



Input

Input controls the input gain and is performed after the **Auto-input** feature. As such this is rarely useful.



Maximizer

Maximizer is a powerful limiter-based maximizer module, which lets you increase loudness of the audio, so that it penetrates the mix better. However please note that for punchy sounds (used in metal for example) the key is to get the initial transients to penetrate the mix and the maximizer may naturally tame these transients as their level is too high compared to the body of the actual drum. So in these cases use this control cautiously and potentially use **Output** control instead. That would make the output exceed 0dBFS, but that shall be fixed later when all the tracks of the drumset are mixed and glued together.



Saturator

Saturator is a powerful analogue style saturation unit, which lets you add some mojo and increase loudness on the way. Be aware that similarly to the **Maximizer** this may tame the wanted transients in more punchy styles, so use this control cautiously.



Output

Output controls the output gain. By default the output will be tuned to reach 0dBFS.



Depth

Depth controls the dry/wet ratio of all of the compressors at once. Set it to minimum to disable all compressors, which is however rarely useful for parallel bus processing.



Glue

Glue controls the amount of the gluing compression based on specific a analog compressor simulation with this specific type of sound.



Depth

Depth controls the dry/wet ratio of the reverb signal. Set it to minimum to disable the reverb completely.



Length

Length controls the length of the reverb tail. Use a longer tail if there is more "space to fill".



Predelay

Predelay controls the delay of the reverb signal. It is useful to make the signal "longer" using the reverb, even if the reverb is actually very short.



Low-cut

Low-cut lets you remove the lowest frequencies from the reverb signal. This is often useful to produce cleaner mixes.



High-cut

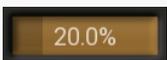
High-cut lets you remove the highest frequencies from the reverb signal. This is often useful to produce cleaner mixes.

Master



Crossover slope

Crossover slope controls the slope of the crossover filters. The higher it is, the less of the high frequencies reach the low band and vice versa.



Warmth

Warmth controls the amount of additional analogue-like saturation used to enrich the audio signal.



Warmth

Warmth controls the amount of additional analogue-like saturation used to enrich the audio signal.

30.0%

Character

Character lets you give the band additional warmth using several analog style compressors. The higher the value, the more compression will be produced. Set it to minimum to disable that.

30.0%

Character

Character lets you give the band additional warmth using several analog style compressors. The higher the value, the more compression will be produced. Set it to minimum to disable that.



Dynamic

Dynamic controls to which extent are the filters dynamic. When set to 0%, the EQ becomes completely static, like if you'd use a traditional parametric equalizer. Increasing this makes the filter move with the sound. 100% makes the EQ fully dynamic, meaning that when the input is silent, it has no effect, but once the level exceeds a certain threshold, the filter starts doing its job. This means that the filter will be moving only when the sound in each band is actually present. This is often very advantageous, since it makes the output much cleaner, not amplifying frequencies longer than actually needed.



Depth

Depth controls the amount of EQ processing.

30.00 Hz

Low-cut

Low-cut lets you use a high-pass filter to remove the lowest frequencies, which could otherwise be causing problems in the mix.

AUTO
INPUT

Auto-input

Auto-input is the first thing to do with the plugin. Run the audio through the plugin, click the button to enable the analysis, wait a few seconds and disable it. It will automatically adjust the internal input gain to make sure the gain staging in the plugin is ideal. If the device has **Matching** feature, this will also produce the spectrum profile for that to work.



Drive

Drive controls the input gain. Since the signal is going through several compressors and other dynamics processors, this in effect controls the amount of processing, hence it is called Drive.



Maximizer

Maximizer is a powerful limiter-based maximizer module, which lets you increase loudness of the audio, so that it penetrates the mix better. However please note that for punchy sounds (used in metal for example) the key is to get the initial transients to penetrate the mix and the maximizer may naturally tame these transients as their level is too high compared to the body of the actual drum. So in these cases use this control cautiously and potentially use **Output** control instead. That would make the output exceed 0dBFS, but that shall be fixed later when all the tracks of the drumset are mixed and glued together.



Saturator

Saturator is a powerful analogue style saturation unit, which lets you add some mojo and increase loudness on the way. Be aware that similarly to the **Maximizer** this may tame the wanted transients in more punchy styles, so use this control cautiously.



Output

Output controls the output gain. By default the output will be tuned to reach 0dBFS.

MB GLUE

MB Glue

MB Glue is multiband (not only) compression module designed specifically for drums. It has "just" 2 bands, which allows processing the problematic low-end separately from the rest. Its purpose is to glue individual elements of the drumkit together and produce a big phat sound we all are seeking. It incorporates several analog-inspired compressors & saturators to achieve that.



Depth

Depth controls the amount of compression performed by the individual compressors inside the MB Glue.



Drive

Drive controls the amount of compression for some initial compression stages. Increase it to produce a bigger sound, lower it to get more natural sound.



Split

Split controls the middle frequency between low and high bands. You may want to change it if your bass drum is tuned very high/low for example.



Bass

Bass controls the low frequency band processing. You can mute it by disabling it. Bass band usually contains mostly the bass drum and low toms and is a subject for problems in the mix. Its audio level is also usually very high, making it difficult to process the drumset together without a crossover.



High

High controls the higher frequency band processing. You can mute it by disabling it. High band contains most of the drumkit, hence this is what you should mainly be focusing on.



Glue

Glue provides a broadband compression for gluing up elements of the drum kit, It can be used as an alternative to **Glue MB**, or use both at once. It employs several analogue-style compressors used in series. Each has a different character and you can control the amount of processing for each. This method produces very natural results as opposed to using just a single compressor.



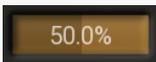
Depth

Depth controls the amount of compression performed by the Glue module.



Warmth

Warmth controls the amount of additional analogue-like saturation used to enrich the audio signal.



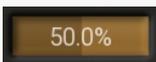
MSL

MSL controls the amount of compression performed by the individual analogue compressor simulator.



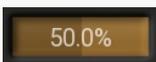
Meve

Meve controls the amount of compression performed by the individual analogue compressor simulator.



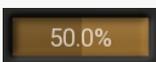
1176

1176 controls the amount of compression performed by the individual analogue compressor simulator.



DBMeld

DBMeld controls the amount of compression performed by the individual analogue compressor simulator.



Stress

Stress controls the amount of compression performed by the individual analogue compressor simulator.



EQ

EQ controls the mastering equalizer, which you can use to shape the sound of the drumkit.

REVERB

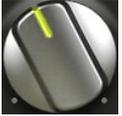
Reverb

Reverb provides an advanced algorithmic reverb chosen specifically for processing this particular type of sound. It may be useful to make the sound wider and bigger. However, note that the main effect of spaciousness should come from the room and overhead mics.

Mode



Mode lets you choose from several algorithmic reverb models specifically selected for this type of sound. The modes are usually sorted from the smallest to the biggest and most audible.



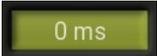
Dry/Wet

Dry/Wet controls the ratio between the dry and reverberated signals. The range is intentionally limited to 40% as that seems a useful maximum for most needs with drums.



Length

Length defines the reverberation signal length. In other words, how long the reverb will become inaudible after the drum hit. In effect, this makes the virtual space smaller or bigger.



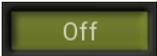
Predelay

Predelay defines the initial delay before the actual response, which simulates the space between the sound source and the listener. The longer the predelay is, the further away the source seems. At some point (around say 100ms) the brain stops understanding that the reverberation belongs to the dry signal and starts interpreting them separately, and then the dry signal becomes close to the listener again. Predelay can therefore be used to control the distance from the source to the destination, but detaching the 2 signals can also be useful for instance to fill up a mix that isn't full enough without smearing the signal with the reverb itself.



Low-cut

Low-cut lets you remove the lowest frequencies from the reverb signal. This is often useful to produce cleaner mixes.



High-cut

High-cut lets you remove the highest frequencies from the reverb signal. This is often useful to produce cleaner mixes.



Time graph

Time graph button switches between the metering view and the time-graphs. The metering view provides an immediate view of the current values including a text representation. The time-graphs provide the same information over a period of time. Since different time-graphs often need different units, only the most important units are provided.



Pause

Pause button pauses the processing.



Popup

Popup button shows a pop-up window and moves the whole metering / time-graph system into it. This is especially useful in cases where you cannot enlarge the meters within the main window or such a task is too complicated. The pop-up window can be arbitrarily resized. In metering mode it is useful for easier reading from a distance for example. In time-graph mode it is useful for getting higher accuracy and a longer time perspective.



Enable

Enable button enables or disables the metering system. You can disable it to save system resources.



Collapse

Collapse button minimizes or enlarges the panel to release space for other editors.



Collapse

Collapse button minimizes or enlarges the panel to release space for other editors.



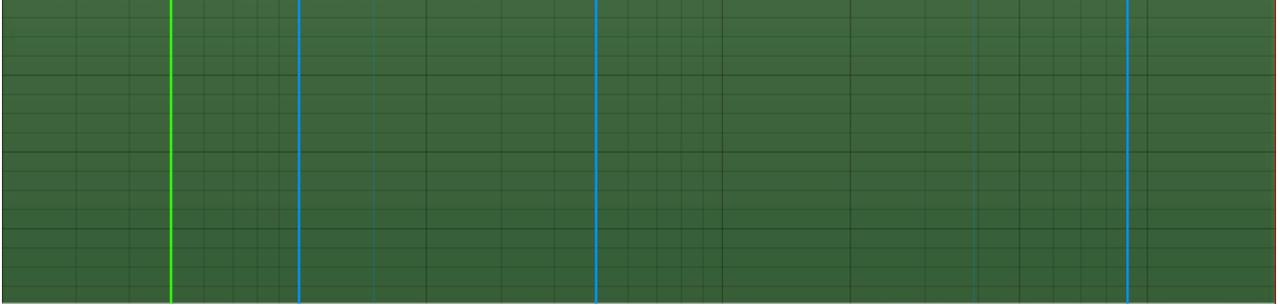
Collapse

Collapse button minimizes or enlarges the panel to release space for other editors.

Used controls

Here we discuss the general properties of all application controls. As a most important rule you should note, that you can always use any question mark button or F1 (or Ctrl+F1 or Ctrl+H) key with the mouse cursor over a specified control to get detailed information about what it does and how to use it.

Graph editor



Graph editor will show and edit one or more graphs.

Installation, activation, introduction to audio plugins

Installation

All MeldaProduction plugins are currently available for Windows and Mac OS X operating systems, both 32-bit and 64-bit versions. You can download all software directly from our website. Since the installation procedures for the two operating systems are quite different, we will cover each one separately.

The download files for the effects include all the effects plug-ins and MPowerSynth. During the installation process you can select which plug-ins or bundles to install. If you have not licensed all of the plugins in a bundle then you just need to activate each plugin separately.

If you have multiple user accounts on your computer, always install the software under your own account! If you install it under one account and run it under a different one, it may not have access to all the resources (presets for example) or may not even be able to start.

Installation on Windows

All plugins are available for VST, VST3 and AAX interfaces. The installer automatically installs both the 32-bit and 64-bit versions of the plugins.

Note: Always use 32-bit plugins in 32-bit hosts, or 64-bit plugins in 64-bit hosts. 64-bit plugins cannot work in 32-bit hosts even if the operating system is 64-bit. Conversely, never use 32-bit plugins in 64-bit hosts. Otherwise they would have to be 'bridged' and, in some hosts, can become highly unstable.

You can select the destination VST plugins paths on your system. The installer will try to detect your path, however you should check that the correct path has been selected and change it if necessary. In all cases it is highly recommended to use the current standard paths to avoid any installation issues:

32-bit Windows:

C:\Program files\VstPlugins

64-bit Windows:

C:\Program files (x86)\VstPlugins *(for 32-bit plugins)*

C:\Program files\VstPlugins *(for 64-bit plugins)*

If your host provides both VST and VST3 interfaces, VST3 is usually preferable. If a plugin cannot be opened in your host, ensure the plugin file exists in your VST plugin path and that if your host is 32-bit, the plugin is also 32-bit, and vice versa. If you experience any issues, contact our support via info@meldaproduction.com

Installation on Mac OS X

All plugins are available for VST, VST3, AU and AAX interfaces. Installers create both 32-bit and 64-bit versions of the plugins.

If your host provides multiple plugin interface options, VST3 is usually preferable. If you experience any issues, contact our support via info@meldaproduction.com

Most major hosts such as Cubase or Logic should work without problems. In some other hosts the keyboard input may be partly non-functional. In that case you need to use the virtual keyboard available for every text input field. You may also experience various minor graphical glitches, especially during resizing plugin windows. This unfortunately cannot be avoided since it is caused by disorder in Mac OS X.

Uninstallation on Windows

The Uninstaller is available from the Start menu and Control panel, in the same way as for other applications. If you don't have any of these for any reason, go to Program files / MeldaProduction / MAudioPlugins and run setup.exe.

Uninstallation on OSX

The Uninstaller is available from Applications / MeldaProduction / MAudioPlugins / setup.app.

Deleting all data, presets etc.

Even if you uninstall the plugins, some data will be left behind - because of potential crossdependencies or because these are your presets, settings, configurations etc. If you want to wipe out everything, please manually delete following folders:

Windows:

C:\ProgramData\MeldaProduction

C:\Users\{username}\AppData\Roaming\MeldaProduction

OSX:

Macintosh HD/Library/Application support/MeldaProduction/
HOME/Library/Application support/MeldaProduction

Performance precautions

In order to maximize performance of your computer and minimize CPU usage it is necessary to follow a few precautions. The most important thing is to keep your buffer sizes (latency) as high as possible. There is generally no reason to use latency under 256 samples for 44kHz sampling rates (hence 512 for 96kHz etc.). Increasing buffer sizes (hence also latency) highly decreases required CPU power. In rare cases increasing buffer sizes may actually increase CPU power, in which case you can assume your audio interface driver is malfunctioning.

You should also consider using only necessary features. Usually the most CPU demanding features are oversampling and modulation of certain parameters. You can reduce modulation CPU usage at the cost of lower audio quality in Settings/Settings/Modulator protection.

Troubleshooting

The plugins are generally very stable, there are known problems however.

GPU compatibility

The software uses hardware acceleration to move some of the processing (mainly GUI related) from your CPU (processor) to your GPU (graphics processing unit). It is highly recommended to use a new GPU, as it will provide higher performance improvements, and update your GPU drivers. Older GPUs are slower and may not even provide required features, so the software will have to perform all calculations in the main CPU. We also have had extremely bad experiences with GPUs from ATI and despite the fact that software is now probably bulletproof, it is recommended to use NVidia GPUs as there has not been a single case of a problem with them.

If you experience problems with your GPU (crashing, blank/dysfunctional GUI), and that you cannot disable the GPU acceleration from the plugin's Settings window itself, download this file:

<http://www.meldaproduction.com/download/GPU.zip>

And place the GPU.xml included in the zip into

Windows: C:\Users\{username}\AppData\Roaming\MeldaProduction
Mac OS X: ~/Library/Application support/MeldaProduction

Memory limits of 32-bit platform

Most hosts are now 64-bit ready, however some of them are not or users willingly choose 32-bit edition, because the required plugins are not 64-bit ready yet. All our software is 64-bit ready. Please note that you must NOT use the 64-bit plugins in 32-bit hosts, even if you have a bridge. If you are stuck with a 32-bit host for any reason, note that there is a memory limit (about 1.5 GB), which you may not exceed. This can happen if you load too many samples or different plugins for example. In that case the host may crash. There is no other solution than to use a 64-bit host.

Updating

You can use "Home/Check for updates" feature in any of the plugins. This will check online if there is a newer version available and open the download page if necessary.

To install a newer (or even older) version you simply need to download the newest installer and use it. There is no need to uninstall the previous version, the installer will do that if necessary. You also do not need to worry about your presets when using the installer. Of course, frequent backup of your work is recommended as usual.

Using touch-screen displays

Touch screen displays are supported on Windows 8 and newer and the GUI has been tweaked to provide a good workflow. Up to 16 connections/fingers/inputs are supported. Any input device such as touch-screens, mouse, tablets are supported. These are the main gestures used by the plugins:

- Tap = left click
- Double tap = double click
- Tap & hold and quickly tap next to it with another finger = right click. Tap & hold is a classic right-click gesture, however that doesn't provide a good workflow, so came up with this method, which is much faster and does not collide with functionality of some elements.

Purchasing and activation

You can purchase the plugin from our website or any reseller, however purchasing directly from our website is always the quickest and simplest option. The software is available online only, purchasing is automatic, easy and instant. After the purchase you will immediately receive a keyfile via email. If you do not receive an e-mail within a few minutes after your purchase, firstly check your spam folder and if the email is not present there, contact our support team using info@meldaproductio.com so we can send you the licence again.

To activate the software simply **drag & drop the licence file onto the plugin**. Unfortunately some hosts (especially on Mac OS X) either do not allow drag & drop, or make it just too clumsy, so you can use Home/Activate in any of the plugins and follow the instructions. For more information about activation please check the [online video tutorial](#).

You are allowed to use the software on all your machines, but only you are allowed to operate the software. The licences are "to-person" as defined in the licence terms, therefore you can use the software on all your computers, but you are the only person allowed to operate them. MeldaProduction can provide a specialized licence for facilities such as schools with different licence terms.

Quick start with your host

In most cases your host will be able to recognize the plugin and be able to open it the same way as it opens other plugins. If it doesn't, ensure you did installation properly as described above and let your host rescan the plugins.

Cubase

Click on an empty slot (in mixer or in track inserts for example) and a menu with available plugins will be displayed. VST2 version is located in MeldaProduction subfolder. However VST3 version is recommended and is located in the correct folder along with Cubase's factory plugins. For example, dynamic processors are available from the "Dynamics" subfolder.

To route an audio to the plugin's **side-chain** (if it has one), you need to use the VST3 version. Enable the side-chain using the arrow button in the Cubase's plugin window title. Then you can route any set of tracks into the plugin's side-chain either by selecting the plugin as the track output or using sends.

To route **MIDI** to the plugin, simply create a new MIDI track and select the plugin as its output.

Logic

Choose an empty insert slot on one of your audio tracks (or instrument tracks for example) and select the plugin from the popup menu. You will find it in the Audio Units / MeldaProduction folder.

To route an audio to the plugin's **side-chain** (if it has one), a side-chain source should be available in the top of the plugin's window, so simply select the source track you want to send to the plugin's side-chain.

To route **MIDI** to the plugin, you need to create a new Instrument track, click on the instrument slot and select the plugin from AU MIDI-controlled Effects / MeldaProduction. The plugin will receive MIDI from that track. Then route the audio you want to process with the plugin to this track.

Studio One

Find the plugin in the Effects list and drag & drop it onto the track you would like to insert the plugin to.

To route an audio track to the plugin's **side-chain** (if it has one), first enable the side-chain using the "Side-chain" button in the Studio One's plugin window title. Then you can route any set of tracks into the plugin's side-chain from the mixer.

To route **MIDI** to the plugin, simply create a new MIDI track and select the plugin as its output.

Digital performer

In the Mixing Board, find an empty slot in the track you would like to insert the plugin to. Click on the field and select the plugin from the effects list.

To route an audio track to the plugin's **side-chain** (if it has one), choose the track you want to send using Side-chain menu, which appears at the top of the DP's plugin window.

To route **MIDI** to the plugin, simply create a new MIDI track in the Track view and select the plugin as its output.

Reaper

Click on an empty slot in the mixer and a window with available plugins will be displayed. Select the plugin you want to open by double clicking on it or using Ok button.

It is highly recommended to select all MeldaProduction plugins in the plugin window the first time you open it, click using your right mouse button and enable "Save minimal undo states". This will disable the problematic Undo feature, which could cause glitches whenever you change certain parameters.

To route an audio track to the plugin's **side-chain** (if it has one), click on I/O button of the side-chain source track in the mixer. Routing window will appear, there you click "Add new send" and select the track the plugin is on. In the created send slot select the channels (after the "=>" mark) for the send, in stereo configuration 3/4 for example. Note that this way the whole track receives the side-chain signal and all plugins with it. It is possible to send it to a single plugin only, but it is more complicated, please check the Reaper's documentation about that.

To route **MIDI** to the plugin, create a new MIDI track and do the same thing as with side-chain, except you don't need to change output channels.

Live

In Session view, select the track you would like to insert the plugin to. At the left top of Ableton Live's interface, click on the Plug-in Device Browser icon (third icon from the top). From the plug-ins list choose the plugin (from MeldaProduction folder), double click on it or drag & drop it into the track.

The X/Y grid usually doesn't provide any parameters of the plugin. This is because the plugins have too many of them, so you have to select them manually. Check Live's documentation for more information.

To route an audio to the plugin's **side-chain** (if it has one), select the track you want to send to the side-chain and in the 'Audio To' menu, choose the audio track that has the plugin on it. Then in the box just below that select the plugin from the menu.

NOTE: Live does NOT support any interface correctly, it doesn't use the reported buses properly, hence it doesn't work with surround capable plugins. Therefore you need to use VST version, which reports only stereo capabilities by default.

To route **MIDI** to the plugin, create a new MIDI track and in the 'MIDI to' menu, choose the audio track that has the plugin on it. Note that in Live only the first plug-in on any track can receive MIDI.

ProTools

In the mixer click an empty slot to insert the plugin to and select the plugin from the tree. The plugin may be present multiple times, once for each channel configuration (mono->stereo etc.). As of now ProTools do not arrange them in the subfolders, which is a workflow dealbreaker, but we cannot do anything about it. The huge empty space on top of each plugin window, which occupies so much of the precious display area, is part of ProTools and every plugin window and again we cannot do anything about it. In some cases you may experience CPU overload messages, in which case please contact Avid for support. Note that ProTools 10 and newer is supported. RTAS compatibility for PT9 and older will never be added.

To route an audio to the plugin's **side-chain** (if it has one), open the plugin, click on the *No key input* button in the plugin title and select the bus you want the audio taken from. You might need to remember the bus number, unless your ProTools version supports bus renaming. ProTools doesn't support stereo (or surround) side-chains at all.

To route **MIDI** to the plugin, create a new MIDI track and in the mixer click the output field for that track and select the plugin, which should already be in the menu.

FL Studio

First make sure plugins are scanned, either a full scan through the Plugin Manager or an automatic fast scan when you open the Plugin Database section of the browser in FL. The scanned plugins will show up in the Plugin Database > Installed section of the FL browser. The Effects and Generators sections in the Plugin Database will show all "favorite" plugins. These can be checked and unchecked in the Plugin Manager or added in some other ways. These favorites also show up in the Add menu, the menu for the "+" button in the channel rack, when you right click an existing channel button to replace or insert, in the plugin slot menu in the mixer and in the plugin picker (F8). The menus with favorite plugins also have a "More" choice that will show all scanned plugins. The full explanation is in our help file, on the page [Installing Plugins](#).

To route an audio to the plugin's **side-chain**, first set up the mixer: make sure the track you want to receive audio from is sent to the track the plugin as a sidechain ([help](#)). Then set up the plugin wrapper: choose the desired input on the **Processing tab** of the wrapper options.

To route **MIDI notes** to the plugin, first configure the sender: choose a MIDI port for the input device in the MIDI settings (for a hardware device), or an output port in the **wrapper options** (for a VST plugin that produces MIDI). For the receiving plugin, set the input port in the wrapper options to the same value you chose in step 1.

To route **MIDI controllers**, the procedure is different. The usual method in FL is to link CC messages to plugin parameters ([help file](#)). VST plugins will also have 128 CC parameters published (through the wrapper) that can be linked this way. Those will send the specified CC MIDI message to the plugin, instead of changing a published parameter.

GUI styles, editor modes and colors

MeldaProduction plugins provide a state of the art styling engine, which lets you change the appearance to your liking. The first time you run the plugins a style wizard will appear and let you choose the style and other settings. It may not be available in ProTools and other problematic hosts.

By default each plugin has a certain color scheme, which differs based on what kind of plugin is that. Also, sections of some plugins are colorized differently, again, based on what kind of section is that (this can be disabled in global settings). Despite you can change the colors anyhow you want, it is advantageous to keep the defaults as these are standardized and have predefined meaning, so just by looking at a plugin's color you can immediately say what kind of plugin and section is that. Same rules apply when designing devices for easy screens. This is the current set of colors:

Dynamics = orange

Equalization, filtering = green
Reverb, delay = brown/yellow
Modulation = blue
Distortion, limiting = red
Stereo = cyan/yellow
Time, pitch, unison... = purple/pink
Tools = grey

Special colors:

Synchronization = grey
Detection = blue/green
Side-chain = green
Effects = red
Advanced stuff = grey



MELDA production
The only limit is your imagination

About MeldaProduction

The best sound on the market, incredible workflow and versatility beyond your imagination. We create the deepest and the most powerful audio plugins with unbelievable sound and tons of unique features you cannot find anywhere else.

Innovative Thinking

At MeldaProduction, we make the most advanced tools for music production and audio processing. We get inspired by the whole range of tools from the ancient analog gear to the newest digital creations, but we always push forward. We've always felt the audio industry is extremely conservative, still relying on the prehistoric equipment making the job unnecessarily slow and complicated. That's why we invent new technologies, which make audio processing easier, faster, better sounding and more creative.

Sound Matters

In the world full of audiophiles you just need superb audio quality. And that's why we spend so much time perfecting audio algorithms until they sound unbeatable. Everything from dynamic filters to spectral dynamic processing. Our technologies just sound perfect.

Inspiring User Interface

Modern user interfaces must not only be easy and quick to use, but also versatile and the whole visual appearance should inspire you. MeldaProduction plugins provide the most advanced GUI engine on the market. It is still the first and only GUI engine, which is freely resizable and stylable. Our plugins can look as an ancient vintage gear, if you are working on old-school rock music. Or as super-modern

futuristic devices if you are working on modern electronic music.

Easy to Use, Yet Versatile

The only limit is your imagination. Our plugins are with absolutely no doubt the most powerful and versatile tools on the market. Yet we managed to make the plugins easy to use via the devices and smart randomization system. But when you are ready, you are one click away from the endless potential the plugins provide.

Never-Ending Improvements

Most companies create a plugin, sell it and abandon it. We improve our plugins, add features, optimize... until there is nothing left to improve and there are no more ideas. Unfortunately that hasn't happened yet :). And the best thing is that the updates are free-for-life!

MeldaProduction was founded in 2009 by Vojtech Meluzin and is based in Prague, Czech Republic.

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