

USER MANUAL

_BUS FORCE

ARTURIA

_The sound explorers

Special Thanks

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Thank you for purchasing Arturia's Bus FORCE...!

This manual covers the features and operation of the Bus FORCE plug-in.

Be sure to register your software as soon as possible! When you purchased Bus FORCE, you were sent a serial number and an unlock code by e-mail. These are required during the online registration process.

Special Messages

Specifications Subject to Change:

The information contained in this manual is believed to be correct at the time of printing. However, Arturia reserves the right to change or modify any of the specifications or features without notice or obligation.

IMPORTANT:

The software, when used in combination with an amplifier, headphones or speakers, may be able to produce sound levels that could cause permanent hearing loss. DO NOT operate for long periods of time at a high level or at a level that is uncomfortable.

If you encounter any hearing loss or ringing in the ears, you should consult an audiologist.

Introduction

Congratulations on your purchase of Arturia's Bus FORCE...!

Arturia's passion for synthesizers and sonic purity has given demanding musicians the best software instruments for professional audio production.

Arturia also has a growing expertise in the audio field, and in 2017 launched the [AudioFuse](#), a pro studio quality audio interface that features two proprietary DiscretePRO® microphone preamplifiers and a set of top-notch AD/DA converters. This line was recently expanded with the launch of the [AudioFuse Studio](#) and the [AudioFuse 8Pre](#). Arturia has also been busy making effect plug-ins, launching in 2018 the first Arturia effects bundle, which included the [Pre 1973](#), the [Pre TridA](#), and the [Pre V76](#). This was our first step in the field of audio effect processing plug-ins, and a logical one, considering all the experience gathered while programming the effects sections for our own synth plug-ins.

These also allowed us to take a first step in analyzing and reproducing retro and vintage analog audio processing units.

Other bundles followed, dedicated to compressors, filters, delays, and reverbs. More recently, Arturia added [three new effects](#) to the collection, this time dedicated to modulation, this way consolidating its position as a leader in audio effect plug-ins.

The ARTURIA Bus FORCE is part of a set of three new effects plug-ins, dedicated to one of the most important tasks in audio processing: bus processing sound control.

ARTURIA has a passion for excellence and accuracy. This has led us to conduct an extensive analysis of every aspect of some of the most famous and iconic bus and master effects units. We have carefully studied and reproduced the behavior of their electrical circuits and modules.

The result is an effects plug-in that, although inspired by one of the most respected bus and master processing units of the market, includes what may be considered our vision of what a bus and mastering effects unit must be.

Bus FORCE runs as a plug-in in all major formats inside your DAW.

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The Arturia team

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1. WELCOME

1.1. What is a bus effect?

As far as audio effects are concerned, bus or masterbus effects occupy a place of their own. These effects are used to polish a more complex, composite sound, be it a group sub-mix or even the final master bus output. Since their main goal is to perfect and balance the overall sound spectrum (and maybe add some color to it), they are specially tailored and carefully tuned to those tasks.

Therefore, their audio circuits are more meticulously planned, and usually more advanced, their components are chosen with top quality in mind, and more expensive and retro components are used, like real tubes for example.

All in all, what one wants to achieve is a sound that's richer, fuller and with more body, maybe supplemented with the kind of saturation you don't usually get in today's pristine audio chains, where we seek the cleanest sound possible.

Traditionally, there are just a few types of these bus effects: Equalizers, compressors (usually called bus compressors, to differentiate them from the usual channel compressors), and maybe more advanced units with complete audio channels, comprising several modules, as is the case here.



1.2. What about the Bus FORCE?

The Arturia Bus FORCE is a complete bus processing channel tool. This means it has all the chain components usually found in a bus channel. These include (but are not restricted to) equalizer and compressor. In this case, the equalizer and compressor modules are complemented with filter, saturator, and clipping modules.

The distinctive sound of the device this software is modeled after is mostly due to the use of retro components and circuitry. The modular nature of the device allows it to work almost like a mixer, with several parallel buses (dry, compressed and saturated), with or without EQ. This gives the unit enormous flexibility which, complemented with the nature of the analog components, creates a modern classic.

1.3. Where can you use a plug-in like this?

A plug-in like the Bus FORCE is more suitable for polishing and perfecting (and sculpting) complex audio, like auxiliary buses or the master stereo bus. This is what it was made for, and where it can shine. The fact that the plug-in is able to mix several audio paths allows it to act like a small master mixer in itself, with independent channels for the dry path, the compressor path and the saturated path, each with its own volume control and mute button, allowing much appreciated parallel processing.



But you can use this bus channel pretty much anywhere you want. As usual, nothing prevents you from using it as a regular channel tool (as a complete channel processor or using just one or a few of its modules, like Equalizer and Compressor, for example). Used with caution, it can do wonders for the audio. The main audio law applies here: USE YOUR EARS. And "may the FORCE be with you".

2. ACTIVATION AND FIRST START

The Arturia Bus FORCE plug-in works on computers equipped with:

Windows 8 or later and macOS 10.13 or later.

You can use the Bus FORCE plug-in as an Audio Unit, AAX, VST2 or VST3 plug-in (64-bit only).



2.1. Activate the Arturia Bus FORCE license

Once the software has been installed, the next step should be to activate your license, so that you can use it without limitations.

This is a simple process that involves a different software program: the Arturia Software Center.

2.1.1. The Arturia Software Center (ASC)

If you have not already installed the ASC, go to this web page:

<https://www.arturia.com/support/updates&manuals>

Look for the Arturia Software Center at the top of the page, and then download the version of the installer that you need for your system (macOS or Windows).

Follow the installation instructions and then:

- Launch the Arturia Software Center (ASC)
- Log into your Arturia account
- Scroll down to the My Products section of the ASC
- Click the Activate button

That's all there is to it!

2.2. Working with plug-ins

The Bus FORCE can be used in all major digital audio workstations (DAWs), including Live, Logic, Cubase, Pro Tools and others, as it comes in all the main plug-in formats. Unlike with hardware, you can load as many instances of the effect as you find useful. Using the software plug-in has two other big advantages over hardware:

- You can automate numerous parameters using your DAW's automation system;
- Your settings and current plug-in state will be recorded in your project, and you can pick up exactly where you left off the next time you open it.

3. BUS FORCE OVERVIEW

3.1. Arturia's Bus FORCE plug-in

Our goal was to give users the experience of a great tool, a classic with a modern approach of the masterbus channel. The idea of combining multiple modules into one unit came from today's go-to outboard heroes. Manufacturers like Overstayer are known for their multifaceted units that offer the analog brilliance of old-school gear combined with the practicality of a sonic Swiss Army Knife.

Therefore, the plug-in almost acts like a stereo mixer, with three simultaneous audio paths: Dry, compressed and saturated. These can be mixed together, allowing you to obtain the much appreciated parallel processing used by famous producers and engineers worldwide.

This is not only a great bus and mastering tool, it also opens up many sonic creative possibilities, revealing new grooves, color and audio characteristics. Who wouldn't want that available in their DAW?



The user interface we created allies the analog feel of vintage equipment with a modern approach more suitable for working in a computer environment. There are four modules in the lower part of the Main panel, grouped as units in a virtual rack. Each module has its own controls, which look like classic analog knobs. Above these, there is a more visual section, in the right-hand part of which you can turn the modules for each routing path ON and OFF, this way creating many routing possibilities. You also have several VU meters, and Mute buttons for each path. The Mute buttons double (may also work) as Solo buttons.

On the left, you have a context display that can show the audio spectrum overview together with EQ curves or filter curves, compressor curves and gain reduction meters, a graphical representation of saturation, or a graphical representation of the audio path. Finally, on each side, you have the overall audio input and output VU meters.

Finally, in the top part of the Main display, you have Volume Control knobs for the audio input, each of the three audio paths, and the global output.

We will take a detailed look at all of these elements in the [Control Panel chapter \[p.21\]](#). Now, it's time to check out how it works and how it sounds. Let's go!

3.2. Understanding the Bus FORCE Signal Flow

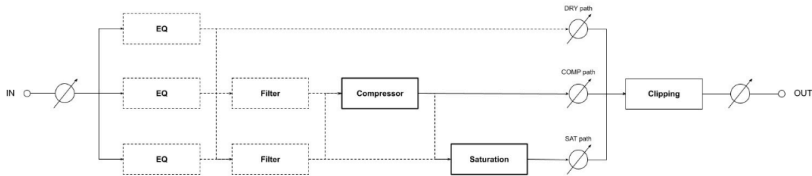
The Bus FORCE is a bus effect plug-in. Look at it as a bus mixer. It will be more useful as an auxiliary bus processor or submix channel bus processor, or even in the master output bus. That's where you will be using the plug-in for the following hands-on tutorial. But first, let's take a look at the inside, to understand a little more about Bus FORCE and its different modules.



Bus FORCE submix and routing display, with the different input choices for each path, the Mute buttons, and the volume controls above. The Saturation module is the only one active in the Sat Path, and the Compressor module is the only one active in the Comp Path.

The plug-in comes in two versions - mono and stereo. Just like the hardware unit that inspired this plug-in, the Bus FORCE has three audio signal paths: Dry, Comp (Compressor) and Sat (Saturation). As a modular unit, the plug-in has several independent "modules", to which the signal can be routed in different ways. Add to this that all three audio paths run completely independently inside the unit and each have a volume control knob, plus a mute button. So that we can say we have a real bus mixer with three independent stereo channels.

But let's have a detailed look at the signal-flow:



When the audio signal enters the unit, it is split into three paths. The first one is the Dry Path. The second one is the Comp Path, which, as the name implies, is routed to the Compressor module. The third one is the Sat path, routed to the Saturation module.

In each of these paths, you have the possibility to activate other modules to further process the signal. The EQ module may be activated in any of the three paths. The equalization setting will always be the same, but you can choose whether you want EQ applied solely to the Dry Path (which will not be completely "Dry" anymore), or to any of the other two paths as well.

The Comp (Compressor) Path can receive the audio directly from the Input (therefore, with no added processing), or from the EQ output. Then you may route the audio through the Filter, or simply turn the filter off, in which case the audio will bypass that module and be directly routed to the Compressor.

Then you have the Sat (Saturation) Path. This path can receive audio signal from the Input, from the EQ output, from the Filter output, and from the Compressor output. When the Compressor output is selected as the Saturation input, the audio comes with all the processing defined for the Comp path, with or without EQ, and with or without Filter. This way, the Compressor can not only feed the final output of the Bus FORCE, it can also feed the input of the Saturation channel. This already gives you an idea of how advanced and flexible this unit is.

There is a final processing module, right after the mixing stage. It's the Clipping module. This is a distortion unit, as the name implies. You have the option to drive the Clipping module with more or less intensity, or simply bypass it.

Add to this that all three audio paths have their own volume control, which means you can have different volume amounts for each signal path in the final output. That's why you can look to the Bus FORCE as a kind of bus mixer.

This makes the Bus FORCE a parallel processing unit, a serial processing unit, and many things in-between. As you can see by now, the possibilities are numerous. As a matter of fact, we have already counted more than 36 possible routing combinations.

And that's it. This is the complete signal flow of the Bus FORCE. It is very flexible, but that flexibility comes with some complexity. Study the circuit carefully, spend some time with the controls of each module, mute and unmute each of the three signal paths, turn each module on and off in each path, try the different routings, and we're sure you'll fall in love with this unit and enjoy its incredible power. The next section will definitely help you with this.

3.3. Getting hands-on with Bus FORCE

3.3.1. The basics

To get an idea of the Bus FORCE's capabilities, we suggest you try the following:

- Load a stereo clip into an audio track in your DAW. A master out or group submix would be ideal;
- Load an instance of Bus FORCE as an insert in that track. Open the Bus FORCE window;
- Ensure the Default preset is loaded. This way, all parameters assume their default values;
- Start playback. You will notice no difference. That's because, with the modules at their default values, the plug-in is not processing the audio in any way. If you look at the volume knobs at the top of the GUI, you'll see that the Dry and Comp paths are muted. Only the Sat Path is working, with all the modules in that path turned on (but in their neutral positions, which means that they will not change the audio);
- You may check this by turning each module in the chain on and off. You can do this in the upper part of the Main panel. When you click a module in the graphical representation of the audio chain, you'll notice that the Visualization Display on the left changes to show the current signal flow diagram. This is handy because it gives you a visual feedback of how the audio is routed;
- Try to unmute the Comp Path. Now turn up the volume control knob. The audio will simply become louder. This is because you now have two audio chains active simultaneously. But nothing else has changed, because all the modules are at their neutral (default) values;
- Right-click the Sat Path Mute button. That button will now be lit up with a blue light, displaying an S. This path is now soloed, which means no other path is sounding. This is a great way to check each channel path's sound without touching anything else, like you do in a mixer (after all, [this is a bus mixer](#)). Right-clicking again will turn Solo off.



The FORCE mixer with the Sat Path soloed and all the modules active. Notice the full colors in module name boxes.

We will now try different routings:

3.3.2. The routings

- Load the Default preset again, to make sure you are working with a clean signal. Routing in Bus FORCE gives you tremendous power in audio processing. With three independent audio paths and the possibility to turn off modules in each path independently, you can probably imagine what we are talking about;
- The initial state only has the Sat path active. You'll notice the Mute button is lit on both the Dry and Comp Paths. On the other hand, all modules are active. You know it because the boxes with the modules names in each path are all in full color;
- Now, unmute the other paths. To do this, simply click the M button. Nothing has changed. This is because not only were the other paths muted, their volume knobs are set to -Infinity. So that, to hear them, you have to turn up their respective volumes. First reduce the Output Trim volume to avoid damaging your ears and your speakers, since the overall volume will become much louder. After all, you will be summing two other audio channels. Double click each volume knob. This will place them at their default values, which is 0 dB;
- Now, let's try a few different routings. This will not have much effect on the audio since all modules are at their default values, which means that, besides adding volume, each chain isn't doing anything to the audio in terms of processing. But this is just to give you an idea of how routing works;
- First, click the Equalizer box in the Dry Path. The box will become fully colored. This means the Equalizer is now On in the Dry Path chain. This is the only module you can activate in this chain. Now click the box with the same name in the Comp Path. The box will become much darker and the name dimmed. This means that the Equalizer, which was On in this path chain, is now Off. The Filter and Compressor modules remain active;
- Now click the Compressor box in the Sat Path chain. You'll notice that the box becomes much darker and the name dimmed, exactly like the Equalizer box in the other chain. The Filter box remains active, though. Let's now try something else. Click the Compressor box again. It is now On;
- Now click the Filter box to turn it Off. Notice that the box becomes dark not only in the Sat Path but also in the Comp Path. This is due to the fact that Saturation is receiving its audio input from the Compressor module, therefore the audio chain has to be the same;
- If the Compressor is inactive in the Sat Path, the Saturation module will receive its audio input from the previous module in the chain instead. That's the Filter module. Now click the Filter box. This time it will only turn Off in the Sat Path: it remains active in the Comp Path. This is because now the Saturation module will receive its audio input from the Equalizer (which is inactive in the Comp Path). This is the routing pictured below;



- To sum up: a module can only be active in one path independently of the other if the higher module in the other path is also inactive in that chain. So, if you have Compressor active in the Sat Path, the whole Comp Path chain will have to be reproduced in the Sat Path chain;
- These are the main rules you have to follow when establishing routings. The example above is just one of many possible routings;

This covers routing. But routing by itself wouldn't do much if you didn't have the possibility to process the audio with each different module. Which you can, of course. And each of these modules is a complete and very powerful audio processor of its own, as you will see:

3.3.3. The EQ

- The best way to try the EQ is to listen to it separately from the other processors. To do this, we will solo the Dry Path. Now the M on the other two channels will be lit, and only the Dry Path will sound in the Output;
- Adjust the Dry Path volume control to fit your desired volume;
- Now adjust the different bands. Notice that, as soon as you touch any control in the EQ section, the display changes to show the EQ curve, and behind it the audio spectrum (input and output). The input spectrum has a darker color, and the outline is dimmed. The output spectrum has a lighter color, and the contour line is brighter. The spectrum is of great help to define the ideal EQ contour;
- We will start with the Presence band. This is the most important one, because it covers the frequencies our ears are more sensitive to. Since our mix lacks a bit of presence, we will raise the band a little. This is a bell band, and you first need to define the center frequency. We want a little more of the mid-high frequencies, so let's turn the Freq knob of this band to approximately 1100 Hz;
- Now, raise the Presence value to 3.00 dB;
- The bandwidth is too narrow for what we want. The low and high bandwidths of the EQ are very soft and wide, based on Pultec-like curves. The mid band is capable of narrower bandwidths, but also very wide ones. Here, we want a bandwidth wider than the default, so we will turn the Width knob to something around 2.7;
- Now that you have set the Presence band, you have to fine-tune the other two. Let's start with the bass band. Since this one also needs some boost, turn up the Low knob to a value around 3.00 dB;
- The default bandwidth is too wide, so you need to narrow it by turning up the Curve knob to a value of approximately 2.3. This is a shelf band, so the frequencies will rise progressively in a soft curve starting a little earlier than the center frequency. Set this center frequency by turning the Freq knob to a roughly 300 Hz. This way, the shelf will start somewhere around 600 Hz, which, coupled with the margins of the bell curve of the Presence band, means that the EQ contour will present a soft valley around that frequency, still above the original volume;
- Since the highs are more or less OK, let's leave that band at the default values;
- Just to hear how the EQ changes the audio, you can now turn the switch Off and On again. When turned Off, the EQ will be bypassed, leaving the audio untouched;



Defining a good EQ contour is a task which requires patience and good ears. If you feel you are becoming tired, stop for a while, rest, refresh your hearing, and then start over.

Now that you see what you can do with the Equalizer, it's time to move on. Next step will be the Filter:

3.3.4. The Filter

- Equalizers usually have filters to cut the extreme highs and lows. Here, the Filter is an independent module, because it is a real synthesizer filter that goes way beyond the simple task of cutting highs and lows. In fact, the low-cut and high-cut filter bands (here labeled the usual HPF and LPF, which stand for High-Pass Filter and Low-Pass Filter) also have Resonance, allowing for more creative uses, like for example "exciting" the frequencies around the cut;
- Since you are using the Bus FORCE as a master bus processor in this hands-on tutorial, you will not use Resonance (the Peak parameter knobs). Bear in mind that these knobs behave pretty much the same way as the Resonance parameter on any synth filter;
- Let's start by cutting the extreme lows. Frequencies below 20 Hz are not heard, yet they transmit a very high acoustic energy, boosting the overall volume without any audible result. So, it is usually a good idea to cut these sub-frequencies;
- The default position for the low-cut (HPF) filter is off. Turn the knob a little. The small display box will read something around 20 Hz. Remember that you can press the Ctrl key or right-click while turning the knobs with the mouse to fine-edit the values. If you haven't yet, press Play to hear the audio. Now, keep turning the knob as long as you feel that the audio is not affected. The sub-frequencies are now being cut without affecting the audible audio;
- You may also want to cut the extreme highs. This may be handy to cut the hiss coming from recordings as well as other artifacts, or just to soften the audio output a little. You definitely want to do it in this mix. Again, the default for the high-cut (LPF) is off. Turn the knob to roughly 12 kHz to soften the audio;
- Remember to take a look at the spectrum display and the filter curve display in the upper part while the audio is playing. This may help you a lot in editing;



Now that we have looked at the filter, it's time to move on to the Compressor:

3.3.5. The Compressor

- The Compressor is one of the main modules in the Bus FORCE. We told you about the importance of compressors when we released our [Compressor Bundle](#). This compressor, together with Comp DIODE-609, which is also part of FX Collection 2. Again, to better listen to the compressor capabilities, put the Compressor path in solo mode. This way, only the Filter and the Compressor will be heard in the final output;
- First, notice the switch in the center, labeled Stereo/Dual. This allows us to treat the audio as full stereo or dual mono. When in dual mono, each channel of the detection signal affects only the corresponding channel of the audio being processed. This may be handy when the stereo image is too wide, as it will compress the loudest channel more and the quieter one less, this way creating a more balanced output. For now, leave it in the default Stereo position;
- Now, let's define the Threshold. This is perhaps, together with the ratio, one of the most important parameters. Since we are processing a final mix, you do not want too low a threshold, but let your ears decide. So press Play and start turning the knob. By default, it is at 0.00, which means it will not compress anything. Since your mix is very low and with strong vocals, you want to compress a little more. So turn the knob down to -24 dB. This is probably more than what you will need in your mix. Once again, remember to look at the display, where you will see the historic volume contour (both Input and Output, as usual), and above these a line showing the gain reduction;
- Now that you have defined the Threshold, it's time to set the compression ratio. This is done by using the Force knob. This parameter is named Force because it isn't just the usual ratio parameter. In fact, Force acts as a multiplier, interacting with the detector mix signal and the release parameter (when it is short enough). This leads to some unusual behaviors, especially when Force value is above 5.00 (marked with an infinity symbol in the knob, because that's when the compression ratio reaches infinity, in practice acting as a Limiter). Above that value, the stronger the detection signal, the lower the audio will play, resulting in volume reduction. Please refer to the [Control Panel chapter \[p.21\]](#) for more details about this very important parameter. For now, we will use the compression ratio. Let's set a value of 3.00, which is soft enough, as the graphic curve shows;
- Now we have to set the Attack and Release parameters. You want an Attack that's not too fast, to let the attack transients pass, but the default Attack (100 ms) is too slow for our purpose. Set an attack value of 75 ms. For the Release, the default value of 30.0 ms seems good enough, so let's keep it;
- Now, we will need to set the Make Up. This allows you to regain some volume after the Gain Reduction performed by the compression circuit. Since we applied some strong compression, we need to compensate, so set it to 6 dB;
- This completes the most important parameters. But there are still some we haven't touched yet. These are the Side-Chain Signal Source selector (which also has a small button to audition it independently, meaning that you can listen to it without processing when it is the internal audio) and the SC Filter, which has two parameters: SC Freq and SC Peak/Gain;

- The SC Signal Source selector allows you to select between Internal and External sources. Usually, the internal audio is used as an SC audio signal for detection purposes, especially in a bus signal. But you can also use an external audio signal (which may be handy if you are using it in an audio channel). Since each DAW handles external side-chains differently, please refer to your DAW's manual to know how to configure external signals as side-chain sources;
- The side-chain signal may be processed independently before being used for detection. You can use the EQ Presence band or the HPF (low-cut) filter. The SC Filter switch has three positions. In the middle position (default), no processing is applied. The left position selects the High-Pass (low-cut) filter, in which case we will use the SC Peak / Gain knob to define the filter resonance gain. The right position selects the Presence EQ band. The SC Freq knob on the left allows you to tune the center frequency for the EQ band or for the Filter;
- These extra processing options allow for some fine-tuning of the detection signal. The Listen button, on the other hand, allows you to listen to the result of the processing;



Now that we have examined the Compressor, it's time to move on to the Saturation module. As with the Compressor, this is a complete and highly versatile module:

3.3.6. Saturation

- Saturation is a kind of distortion (harmonic distortion) that enriches the audio spectrum in a way considered pleasant. In fact, to be perceived as unpleasant, harmonic distortion has to reach very high levels. At lower levels, it is not only bearable, it is desirable. This distortion used to come from tube and tape saturation (hence the name). In today's pristine digital audio environments, there is usually no harmonic distortion. That's why you need extra processors to recreate it;
- This module only has a few controls: Harmonics, Overdrive and Make Up. You already know what Make Up does, so let's take a look at the other two;
- First, let's isolate the Saturation processing path by soloing it. This will allow you to better hear the Saturation effect in the final audio output. We will start with Harmonics. Overdrive is another (different) matter;
- To the right of the control knob, you have a selector. By default, it reads "Thick". Click it. You'll notice that the display in the section above changes (once again). As we've seen, that display is context sensitive, which means it will always change to show what is relevant to the parameters you are currently touching. Now, it is showing the display relevant for Saturation;
- The Saturation path only has the EQ and the Saturation modules active. Since you've set a very gentle EQ, that's an ideal scenario to experience the power of this saturation circuit. When you clicked the Thick button, a context-sensitive menu presented you with three options: Even, Odd and Thick. Even particularly boosts even harmonics (2, 4, and so on) and is usually considered the softest saturation. Odd favors odd harmonics (3, 5, etc.) and sounds a little harder. Thick boosts all of them equally, so this is the option that gets you the strongest saturation. Too much saturation is often not a good idea (unless that's what you really want), so we'll go sparingly for now. Let's stick with the "Thick" option;
- Hit Play. Now, raise the Saturation knob until you find it is becoming unpleasant. Turn it back to the point where it doesn't sound unpleasant anymore. You may try it again once or twice, to find the sweet spot. Once you feel you've gotten it, leave it there;
- If you feel you need to, you may adjust the Make Up gain. Remember this is just one of three audio paths, and you may want to use ALL of them in the end. Be mindful of not using too much Make Up gain;
- Notice the display. In the right part, you have a waveform. When saturation is at 0, the waveform is a perfect sine. The higher you raise the Saturation, the more it looks like a square wave. This helps you get an idea of what the parameter is doing to the audio;
- Overdrive is another kind of distortion. It may sound harder or stronger (remember all these adjectives are highly subjective when applied to audio). This distortion is usually used (and seen) in guitar amplifiers, and guitar players love it. You may want to add a bit of Overdrive to your audio to "brighten" it a little. We'll use 3 dB here;
- The curve switch performs a pre-emphasis of the audio before saturation (you may think of it as a kind of "grit" parameter control). This works by lowering the lows before the audio reaches the saturation stage. As a result, at higher saturation values, mid and high frequencies distort a little more than the lows. This allows you to get the aforementioned "grit" while preserving the definition, body and warmth in the bass frequencies (which most of the time will simply be boosted, but not distorted). By default, it is on, but you have the option to turn it off;



And that's pretty much it as far as the processing modules are concerned. Now, we will continue our journey through the Bus FORCE features by looking at some extras:

3.3.7. Clipping and mixing output

- Clipping is another kind of distortion. Basically, it "squares" the waveform when it goes over zero. As a consequence, it creates a lot of high frequency partials. This may or may not be desirable. For example, in the "noisy" short attack transients of drum sounds, it may be much more effective than a limiter, allowing for a much "punchier" sound. Since the results may be "harsh", it is off by default. But there is always a first time. You may want to try it (like on drums, for example) and evaluate for yourself if you like it or not;
- The upper part of the Main Panel contains the Volume knobs that control each audio path, as well as the Output. The Input volume control is to the immediate right of the Power switch on the left side of the GUI;
- This switch places the plug-in in bypass. When in bypass, the panel becomes darker and dimmed, and the word "bypassed" appears in the center. There is no audio being processed by Bus Force when it is in bypass;
- To the right of the Output Trim knob, there is a Link button. When this button is On, the Input and Output Trim knobs are linked. This means that, if you raise the Input volume, the Output will move down to compensate. This way, you can have a steady Output volume while trying different Input values. The Output knob will always move independently;



These are just a few examples of what you can do with Bus FORCE. This is a very complex and sophisticated audio processor, capable of changes that can be either subtle or drastic, depending on how you set it. It offers the user a lot of processing power, and if you pay attention to detail and invest some of your time to learn how to use it, you will be able to get the most out of it. As always, use your ears and persevere, to find the best and more interesting ways of using it.

You can find a few templates in the presets list, covering all the serial, parallel and multiband routings you may want to start with. These are a great starting point if you already know what you want to accomplish. Please check them out, as they can be really handy.

4. BUS FORCE CONTROL PANEL

The Bus FORCE plug-in can be used in Mono or Stereo configurations.

The Mono configuration is automatically loaded when you use the plug-in with mono tracks. When inserted in stereo tracks, the Stereo configuration is automatically loaded as well.

By default, the plug-in works according to the source material.



⚠ Not all DAWs work with mono tracks, in which case you will not be able to use the mono configuration.

4.1. Channel Configuration (Mono/Stereo)

There are almost no differences between Mono and Stereo configurations except the Stereo/Dual Mono switch for signal detection in the Compressor module, which is clearly of no use in a mono signal.

When instantiated in stereo channels, the plug-in loads in stereo mode.

When instantiated in mono channels, the plug-in loads as a mono plug-in. Although the main panel configuration looks almost exactly the same, only the left channel receives and processes audio.

4.2. Main Control Panel

The Bus FORCE Graphical User Interface (GUI) is a combination of the look of classic hardware units and more modern computer-based displays. We feel this 'hybrid' approach works best in a computer-based graphical environment, without losing the classic 'retro vibe and feel' we love so much.

As all current Arturia plug-ins, this GUI has an Upper Toolbar and a Lower Toolbar. The Lower Toolbar allows you to access the Undo and Redo functions, lists the editing history, allows you to put the plug-in in bypass (the Bypass button mirrors the Power button on the Main control panel) and displays CPU consumption.

The Upper Toolbar is where you access the main menus, perform important tasks like loading and saving presets and banks of presets, and where you can select a preset and see the name of the active preset. The toolbars and their features are covered in detail in the [User Interface chapter \[p.42\]](#).

We'll now have a look at all the controls and discover their function, their ranges, and how to interpret the numbers.



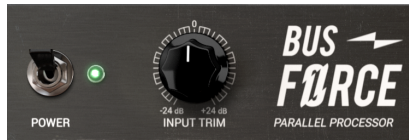
Bus FORCE Main Panel instantiated on a mono channel, with all controls in their default positions. Notice the absence of the switch above the GR LEDs in the Compressor

Notice that each time we click a control (knob or button), or simply hover the mouse over it, the Lower Toolbar displays the parameter name at the bottom left. Also, a small pop-up box displaying the current parameter value appears to the right of the control. This changes every time you move that control, updating the parameter value in real time.

Now, let's take a look at each control on the Main Control Panel.

4.2.1. Power

This switch turns the plug-in on. Turning it off has the same effect as pressing the Bypass button in the Lower Toolbar - it puts the plug-in in bypass mode. When in bypass, the GUI is grayed out and shows "Bypassed" in the middle.



Bus FORCE Power Switch, Input Trim knob and plug-in logo

4.2.2. Input

This controls the level of the input signal. By default, it is positioned at 0.00 dB, which means it has no effect (the input signal passes unaffected). Range goes from -24 dB up to 24 dB.

i ⚠: Driving the signal too hot may lead to distortion, while reducing the level too much may lead to insufficient gain. However, it may be good to reduce input a little if you plan to drive saturation high.

4.2.3. Dry Path

This knob controls the dry path audio circuit. This path can only be processed by the EQ. It has its own volume control and mute button, therefore you can mix it with the other audio paths in variable amounts. The Mute button works as a Solo button when you right-click it.

By default, the Dry Path volume knob is set to 0.00 dB. Its range goes from -Infinity to +12 dB.

4.2.4. Comp [Compressor] Path

This knob controls the compressor audio path volume. Again, this circuit has its own volume control and mute button. The Mute button works as a Solo button when you right-click it.

This is an analog VCA type compressor. It has the usual compressor controls, but it may act a little differently than a linear compressor. It has two detection paths, one reacting to audio peaks, therefore creating more compression and "pumping", and another reacting to audio RMS, which is more stable, therefore acting more like a tube compressor, slower and smoother - a behavior you might consider "more musical".

By default, the Compressor Path volume knob is set to 0.00 dB. Its range goes from -Infinity to +12 dB.



Bus FORCE Dry Path, Comp Path, Sat Path, and Output volume controls. Notice that the Link switch on the right is activated. This means that the Output Trim knob will compensate for any changes applied with the Input Trim knob

4.2.5. Sat [Saturation] Path

This knob controls the saturation audio path volume. This circuit also has its own volume control and mute button. The Mute button works as a Solo button when you right-click it.

This is an analog saturation module with several saturation modes and an independent Overdrive.

By default, the Saturation Path volume knob is set to 0.00 dB. Its range goes from -Infinity to +12 dB.

4.2.6. Output

This knob controls the overall output volume of the Bus FORCE plug-in. As with the previous controls, its range goes from -Infinity to +12 dB. By default, it is positioned at 0 dB.

4.2.7. Link switch

This switch links the Output Trim knob to the Input Trim knob. If the switch is on and you move the Input knob, the Output knob will move in the opposite direction to compensate for the change. That way you can try different input levels while maintaining a constant output level.

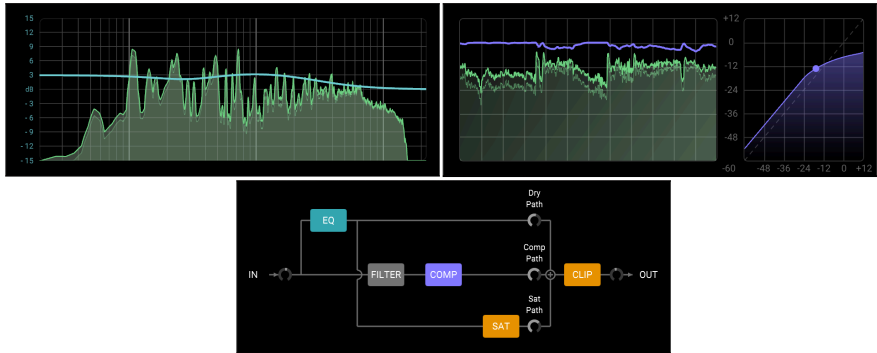
This only links the Output to the Input and does not work the other way around. The Output knob will always move independently.

4.2.8. Visualization Display

The visualization display is located on the upper left side of the Main Panel. We strongly advise you to always pay attention to it, as it displays very important visual feedback which changes depending on the module you are tweaking, or even when you are working on the routings in the upper part of the Main panel. It helps a lot by allowing you to visually control what's happening to the audio.

This kind of visualization was never a possibility in classic analog units, but we feel it greatly improves the user experience and is particularly well-suited to a graphical working environment like the computer.

As we said, the visualization changes depending on the task being performed. We will now list all the different screens that can be displayed:

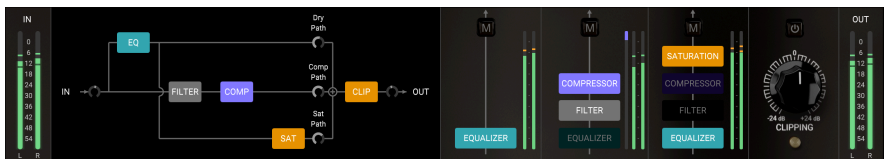


4.2.8.1. Routing display

This visualization is activated anytime you touch any of the mixer controls. This shows a signal flow diagram of the audio, which reflects the current routings of the unit. You will clearly see the different paths, and the modules that are On and Off. Modules that are On are displayed in the path. If a module is completely bypassed, it will simply disappear from the diagram.

Muted, soloed and active paths are also identified. A small graphical ring represents each volume knob.

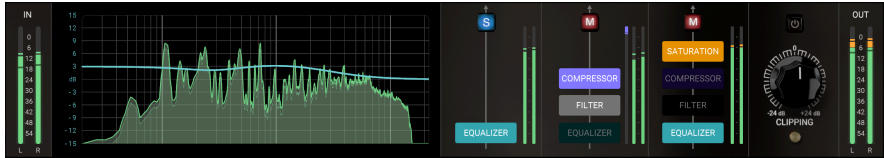
This is just for display. Any changes in routing (like turning modules On and Off for each path, soloing or muting a channel path, etc.) have to be performed in the mixer window on the right.



4.2.8.2. EQ display

As soon as you touch a control in the EQ module, the visualization window immediately changes to the EQ display. This shows the global EQ curve. It also shows the spectrum signal before and after EQ. The spectrum after EQ is shown brighter and at the front, while the spectrum before EQ is shown behind, dimmed and in a darker color.

This allows you to clearly see the changes applied by EQ on the audio. When the Equalizer module is Off (the small switch at the top is in the Off position), the display is dimmed and shows the words "EQ Off".



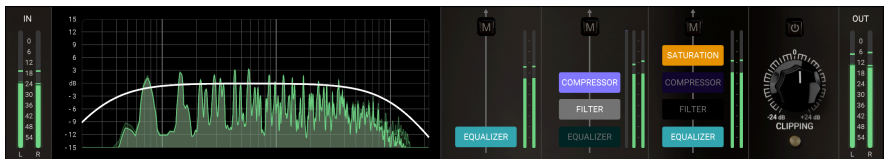
4.2.8.3. Filter display

When you touch any control in the Filter module, the visualization immediately changes to the corresponding display. This is the common behavior so, if you want a certain display in the Visualization window, all you need to do is touch any control of that module.

The Filter display is very similar to the EQ display. It shows the global filter curve. This means that you see a low curve if only the low-cut (HPF) filter is active, a high curve if only the high-cut (LPF) filter is active, or both if both are active. The curve shows the resonance (peak) curve in the filter if that is also active.

Behind the curve, you have the spectrum signal before and after filtering, as with the EQ. The spectrum after filtering is shown brighter and at the front, while the the spectrum before filtering is shown behind, dimmed and in a darker color.

This allows you to clearly see the changes applied by the Filter module on the audio. When the Filter module is off (the small switch at the top is in the off position), the display will be dimmed, showing the words "Filter Off".



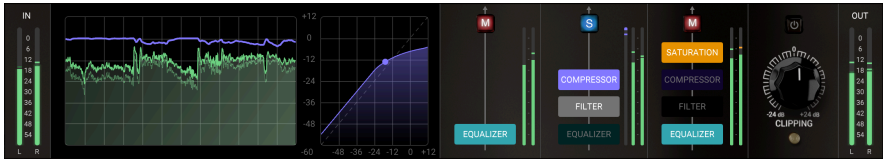
4.2.8.4. Compressor display

Touching any control on the Compressor module immediately activates the Compressor visualization, which is made up of two windows. On the left, you have a spectrum display, this time adapted to compression. This means that you have the historic volume contour of the audio, and above it a line indicating the gain reduction applied by the compressor.

As with the other modules, the spectrum display shows the spectrum before and after compression; the spectrum after compression is shown brighter and at the front, while the the spectrum before compression is shown behind, dimmed and in a darker color.

Then you have the typical compressor curve graph. This is the usual square, where the X axis (horizontal) represents the Input, and the Y axis (vertical) represents the Output. This allows you to clearly see the effect of the compressor ratio and threshold on the audio. It will also show the "original" behavior of the Force parameter (which replaces Ratio in the Bus FORCE compressor), when it goes over 5.00.

When the Compressor module is off (the small switch at the top is in the off position), the display is dimmed, showing the words "Compressor Off".



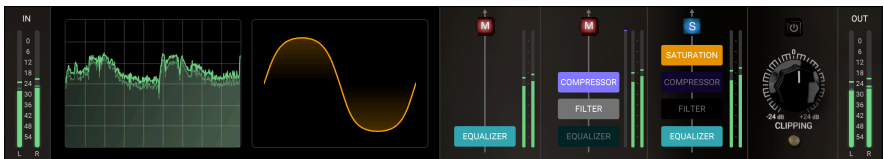
4.2.8.5. Saturation display

Touching any control on the Saturation module immediately activates the Saturation visualization, which is very similar to the Compressor visualization and is also made up of two windows. On the left, you have the spectrum display, this time adapted to saturation. This means that you will only have the historic volume contour of the audio, before and after saturation.

Once again, the spectrum after saturation is shown brighter and at the front, while the spectrum before compression is shown behind, dimmed and in a darker color.

Then you have a graphical representation of saturation. This is represented by a sine wave (no saturation) which progressively becomes "squared" when the saturation (or the overdrive) value is raised. This gives you an idea of what the effect of those two parameters on the audio. Of course, this is just a "graphical representation" with no direct correspondence to what happens with the audio. For that, the spectrum display will give you a better image.

When the Saturation module is off (the small switch at the top is in the off position), the display is dimmed and shows the words "Saturation Off".



4.2.9. Mixer and Routing

To the right of the Visualization display, you have the mixer and routing window. The Bus FORCE is a parallel bus mixer with three channels: Dry, Comp and Sat. Sat receives audio from the Saturation output module. Comp receives audio from the Compressor module. Dry receives audio from the Dry input OR from the Equalizer module.

Each channel allows you to have several modules on or off, independently. For example, you may have the Equalizer on for the Dry Path, but off for the other two paths, or have it on for the Dry and Compressor paths, but off for the Saturation path.

Paths get progressively more complex, as each can have more modules in the chain than the previous one. The Dry path can only have one module active (the Equalizer). The Comp path can have the Equalizer, the Filter, and the Compressor modules (remember that you can deactivate any of these by clicking them in the corresponding path). Finally, the Sat path can have the Equalizer, the Filter, the Compressor and the Saturation modules.

With three audio channels and the possible activation/deactivation of modules in each channel, many routing possibilities are available.

Add to this that each channel has its own volume control (the knobs in the upper part of the Main panel), as is expected in a mixer, and also a Mute/Solo button. These buttons are at the top of each channel and are labeled with an M. When you click one of them, it lights up red. If you right-click it, it lights up blue and its label changes to S (Solo). Consequently, the other channel buttons also light up red, indicating they are muted.

The mixer also has its own VU meters, as would be expected. On the left-hand side of the panel, you have the Input VU meters, while on the right-hand side you have the Output VU meters. The Main Output volume control knob is in the top right corner, after the volume control knobs for each channel.



4.2.10. Clipping

This is the control for the amount of signal that will feed the Clipping module. There's also a button above to turn Clipping on or off.

This control doesn't work as a volume control but as a Threshold control. Therefore, the lower the value (i.e. when you turn the knob to the left), the more clipping effect is applied to the audio. If you turn it to the right, (positive values) it will be less likely for any clipping to be induced, since it will only kick in when the audio goes over the set value.

Clipping is another kind of distortion. It introduces a nastier, harder distortion type and should therefore be used very carefully. Our ears are much less tolerant to this kind of distortion than to harmonic distortion.

This aside, it can still be useful, because the Bus FORCE is not exclusively intended for mastering and bus duties but can also be used for more creative tasks, where that kind of tool can be helpful. You may want to try clipping with drums sounds, as it allows for a much more effective way to get that sought-after 'punch' sound than, say, a Limiter.

The clipping module is placed right below the audio volume output control of the plug-in. You only have one control for clipping (to control the signal threshold where clipping kicks in) and an On/Off switch button. Threshold range goes from -24 dB up to +24 dB, and defaults to 0 dB.

By default, Clipping is off, which means the module is bypassed.

4.2.11. Equalizer

The Equalizer module is an analog type three-band EQ, with curves inspired by the famous Pultec equalizers. The Low and High bands have a shelf type filter, while the Middle band (labeled Presence) has a Peak (Bell) type filter. You have controls for Curve on the Low and High frequency bands, and for the Q (width) in the Presence band. You also have controls for Gain and Frequency, which makes this a fully parametric three-band EQ.

The Equalizer output can be routed to each of the three main audio paths (Dry, Compressor and Saturation), and also to the Filter input. Any or all Inputs may be assigned independently, but the EQ settings will be the same for all of them.



4.2.11.1. Low Shelf Curve

This controls the bandwidth of the High Shelf band. This band uses a Pultec-like EQ curve. The use of this curve means that, even at higher values, the behavior of this EQ will still be softer than that of other EQs. Pultec-like EQ curves are great for shaping tones in a very natural way. Even when frequencies are fully boosted, the boost still has a smooth and natural character to it. The reason for this is the broad nature of the curves featured in those equalizers. Nevertheless, higher bandwidths may have less musical results than lower bandwidths.

Range goes from 0.00 up to 10.0 and defaults to 0.00.

i The famous Pultec low boost may be achieved with a curve between 5 and 10. Find the frequency where the mud is, and the Low Shelf boost will control the mud and boost below, which makes the low end Pultec EQs are known for: huge but not muddy.

4.2.11.2. Low Shelf Gain

This control allows you to boost or attenuate the Low Shelf band frequencies gain. Range goes from -15 dB up to 15 dB, and defaults to 0.

4.2.11.3. Low Shelf Frequency

This controls the center frequency of the Low Shelf band. In the original hardware unit this plug-in is based on, this is a switch with just three options. Here however, it is a fully parametric band control, going from 10 Hz up to 30 kHz (lower and much higher than the usual EQ, reaching regions usually unthinkable for a Low Shelf band). This allows you to use this band way beyond the usual frequency intervals, and for purposes way beyond the typical ones.

Although this gives much more versatility to the EQ, one would expect to use this band in a frequency range around the 100-150 Hz. 150 Hz is the default frequency for the Low Shelf band.

4.2.11.4. Presence Q

This controls the Q factor (frequency bandwidth) of the Presence band. The reason it is called 'Q' goes back to the original days of equalizers when it was originally known as 'Quality'. The bandwidth curves of this EQ are very soft even at high Q values, but even so, higher (narrower) bandwidths may have less musical results than lower (wider) bandwidths.

Range goes from 0.00 up to 10.0 and defaults to 5.00.

4.2.11.5. Presence Gain

This control allows you to boost or attenuate the Presence band frequencies gain. Range goes from -15 dB up to 15 dB, and defaults to 0.

4.2.11.6. Presence Frequency

This controls the center frequency of the Presence band. In the original hardware unit this plug-in is based on, this band has a knob to control the center frequency with an abstract scale from 0 to 10. Although this gives much more versatility to the EQ, one would expect to use this band in a frequency range around 1-2 kHz. As a matter of fact, the default frequency for the Presence band is even lower, and sits at 680 Hz.


Presence is a term that first appeared in guitar amplifiers in the 50s. It also appeared in the most famous Neve equalizers and several graphic equalizers. This term usually refers to a range of mid-high frequencies between 1 kHz and 5 kHz, responsible for clarity and definition in most sound sources. Indeed, human hearing is particularly sensitive to that range and to any changes applied to it.

Cutting frequencies in the presence range may reduce the clarity of an audio signal and cause it to sound distant. On the other hand, boosting this range too much will result in an overly harsh and irritating sound (because of our ears' high sensitivity to these frequencies).

4.2.11.7. High Shelf Curve

This controls the bandwidth of the High Shelf band. This band uses a Pultec-like EQ curve. The use of this curve means that, even at higher values, the behavior of this EQ will still be softer than that of other EQs. Pultec-like EQ curves are great for shaping tones in a very natural way. Even when frequencies are fully boosted, the boost still has a smooth and natural character to it. The reason for this is the broad nature of the curves featured in those equalizers. Nevertheless, higher bandwidths may have less musical results than lower bandwidths.

Range goes from 0.00 up to 10.0 and defaults to 0.00.

 If you want more "air" in your mix without it becoming harsh, you can put the shelf where the high frequencies are dropping, set curve around 4.5, and boost. This will keep everything up to your selected frequency relatively flat, and gradually boost from there. Using curve at 10 may allow 'de-harshing' while increasing details / air. To explore this feature, try boosting at 4K with curve at 10. This will increase high end while reducing the harsh sensitive zone.

4.2.11.8. High Shelf Gain

This control allows you to boost or attenuate the High Shelf band frequencies gain. Range goes from -15 dB up to 15 dB, and defaults to 0.

4.2.11.9. High Shelf Frequency

This controls the center frequency of the High Shelf band. In the original hardware unit this plug-in is based on, this is a switch with just three options. Here however, it is a fully parametric band control, going from 10 Hz up to 30 kHz (higher and also much, much lower than the usual EQ, reaching regions unthinkable for a High Shelf band). This allows you to use this band way beyond the usual frequency intervals, and for purposes way beyond the typical ones.

Although this gives much more versatility to the EQ, one would expect to use this band in a frequency range around 3-10 kHz. 3.9 kHz is the default frequency for the Low Shelf band.

4.2.12. Filter

This module is a synthesizer-like State-Variable Filter, capable of resonance up to self oscillation. This is actually a double filter, with both High-Pass and Low-Pass configurations (and can also perform as Band-Pass, if both filters are engaged).

You have controls not only for Cutoff, but also for Resonance (labeled Peak). The filter can receive its audio input directly from the Dry audio input or from the Equalizer. Since the filter always has to be routed to one of the two main audio paths (Compressor and/or Saturation) and the EQ can also be assigned to each of these, and also to the Dry Path, this routing choice is a kind of parallel pre-routing stage.



4.2.12.1. Input Selector

This is where you select the input source for the Filter module. You can choose to get the audio directly from the Input, or from the EQ output. By default, the Filter receives the audio from the EQ output.

4.2.12.2. HPF Peak

Peak is another name for the parameter usually called resonance. This parameter boosts the frequencies around the cutoff point.

Peak values go from 0.00 to 10.0. Default value is 0.00 (no resonance).

4.2.12.3. High-Pass Frequency

This controls the cutoff frequency. The Peak parameter will boost the frequencies around this point. All frequencies below the cutoff point will be cut with a ratio of 12 dB per octave (which means that frequencies one octave below the cutoff point will be attenuated by -12 dB). This is soft enough to allow the filter to be used musically in any kind of material.

4.2.12.4. LPF Peak

Peak is another name for the parameter usually called resonance. This parameter boosts the frequencies around the cutoff point.

Peak values go from 0.00 to 10.0. Default value is 0.00 (no resonance).

4.2.12.5. Low-Pass Frequency

This controls the cutoff frequency. The Peak parameter will boost the frequencies around this point. All frequencies above the cutoff point will be cut with a ratio of 12 dB per octave (which means that frequencies one octave above the cutoff point will be attenuated by -12 dB). This is soft enough to allow the filter to be used musically in any kind of material.

4.2.12.6. Band-Pass Filter

By engaging both the High-Pass and Low-Pass filters, you can obtain a Band-Pass filter. The effect of the Band-Pass depends on the parameters set for both the High-Pass and the Low-Pass.

4.2.13. Compressor

This module is a VCA type compressor, inspired by a famous modern vintage bus and mastering stereo channel unit. It can go from very subtle ratios, suitable for mastering purposes, to higher, more drastic ratios, more suited to 'creative compression' and sound design. It also has an 'unusual' behavior at higher compressor values.

Being a VCA type compressor, it has controls for all the usual parameters, like ratio (here labeled Force, for reasons we will explain below), threshold, attack, release, make-up gain, and even side-chain (with a choice for side-chain source, and side-chain gain and frequency controls).

The Compressor module may take its audio input from the Dry output, from the EQ output, or from the Filter output (with or without EQ). This means four different alternative input sources for the Compressor path alone.



4.2.13.1. Stereo/Dual Mono

This allows the module to receive the audio as a stereo signal or as a dual mono signal. When in dual mono, the two channels are treated as two completely independent sources, while in stereo, the audio is treated as a single stereo source. This is important because, in dual mono, each channel is compressed differently, as detection is controlled by its own channel signal, while in stereo the detection follows and is controlled by the average volume contour of the two channels.

Generally, in mastering or on the main bus and subgroups, it is a good idea to use stereo mode to avoid 'image shifting' of the stereo signal.


4.2.13.2. Attack

This controls how fast the Compressor starts to act (to reduce the gain) once the Threshold frequency is reached. This is a very important parameter. Faster attacks may lead to the loss of attack transients, while a slightly slower attack time allows the audio to preserve its original attack characteristics, leading to more natural and pleasant results.


Attack range goes as low as 1.00 ms (very, very fast), and may be raised up to 300 ms. Default values is 100 ms, which is slow enough to allow all the attack transients to pass uncompressed.

4.2.13.3. Release

This controls the time it takes for the compressor to release the audio (to stop compression). Usually, release times are slower than attack times, to prevent audible artifacts caused by the sudden transition from compressed audio to uncompressed audio. However, this isn't always the case. On a bus, attack times are sometimes slower than release times. That's what happens with the default values we set here.

 In a mastering scenario, long attack times usually work well with a shorter release, and short attack times with a slower release time.

Range goes from 1.00 ms up to 500 ms. Default value is 30.0 ms, which is a good compromise.

 The 1 ms release time is awesome paired with slow attack times, very low threshold and very high Force values. This leads to 'extreme' results. Feel free to try this and judge for yourself. These settings work well when blended with the Dry Path, resulting in what is known as 'parallel compression'.

Release also interacts with detection. Detection is performed by both Peak and RMS signals. If Release time is low enough, the signal peaks will have a stronger influence in detection than RMS. As Release time increases, RMS gains more influence.

4.2.13.4. Threshold

Threshold is the parameter that sets the volume at which the compressor starts to act (reduce gain). At 0.00 dB, the compressor will not have any effect at all (all audio will pass uncompressed). The lower you set the threshold, the more the audio will be processed.

Signal detection is performed by both Peak and RMS audio signals. Detection performance is quite complex as the relationship between both signals changes following mainly the values of the Force parameter, but also that of the Release parameter. Therefore, Threshold detection may act differently depending on those values.

Threshold range goes from -50 dB up to 0 dB. 0 dB is the default value, which means no compression.

4.2.13.5. Force

This parameter may be seen as a kind of compressor ratio. However there are some things you have to know about it.

Force strongly magnifies the effects of the compression (gain reduction) by a multiplicative factor. The envelope parameters (Attack and Release) are also emphasized, but in a more subtle way.

As the control interacts with the Attack and Release parameters, you can obtain a large array of envelope shaping and sonic possibilities, depending on the selected value and the Attack/Release values. The Force parameter also has other atypical effects for the more adventurous.

In fact, the compressor ratio starts with a soft 1.1 to 1 (at 0.00 value). It reaches infinite ratio at about half value (around 5.00). From that value upwards, the ratios become negative.

Negative ratios mean that the sound under the threshold remains "untouched" (as usual), but the louder a sound gets above that, the more its gain is reduced (to a value lower than the initial value, not just attenuated as is the common compressor behavior). In a way, with force at max, the sounds that will sound the loudest are the ones which are not "touched" by the compressor (the ones under the threshold value).

A better way of understanding what it does is to start with a value of 0 and to raise it while listening to the results. It can create very extreme results at higher values because, on top of the compression ratios becoming negative, the Attack and Release times are multiplied (fast becomes faster, etc.). Bear in mind that the effect of the Force parameter on Attack and Release is more subtle than the effect on the Gain Reduction parameter.

What you should use as a rule of thumb is that the less Force, the more gentle the GR curve is (ratio gets lower, and action is smoother due to a stronger presence of RMS signal in detection), but at the same time, small Release values let the algorithm get a bit more aggressive than it should, because detection relies more on audio peaks than on the RMS signal.

The visualizer may help you understand the effect of the Force parameter on the compression profile more easily.

Range goes from 0.00 up to 10.00. From around 5.00 upwards, the Force parameter stops compressing the usual way (at this value, the compressor effectively acts as a limiter) and starts to act as a "negative" compressor.

4.2.13.6. Make Up Gain

Make Up is how you compensate for the loss of volume due to gain reduction. This is the reason why many people think compressors allow the audio to become louder. As a matter of fact, compressors have exactly the opposite effect. They lower the higher volumes (volumes above the threshold), this way softening the overall audio volume.

That's where the Make Up parameter comes into play. With it, you can compensate the gain loss by boosting the compressed audio volume. As a result, the regions that were lowered may sound as loud as before, but the ones that were not compressed (which didn't suffer any gain reduction) will sound louder. The result is an overall louder sound.

Make up goes from 0.00 dB up to 30.0 dB. Default is 0.00 dB (no amplification)

4.2.13.7. SC [Side-Chain] Source Selector

This is a very important parameter. It selects the audio source the Compressor will react to (the audio signal which will control the Compressor gain reduction). By default, it is set to internal, which means the Compressor will react to the audio input of the track it is instantiated on. However, you can select an external audio source. This way, you can control the GR from another audio source, (for example, a kick or a bass line). This isn't usually the case when the compressor is used for bus and mastering purposes, but SC is often used for compression on single tracks.

Please refer to your own DAW's manual to learn how to configure side-chain.

The way the detection signal works on the Gain Reduction also changes when Dual Mono is selected instead of Stereo. When in Stereo, both channels have the same GR applied, as the amount comes from the average level of the two channels. When in Dual Mono, each channel is treated independently. This leads to different results, depending on how much stereo separation exists in the two stereo channels.

4.2.13.8. Headphones [Listen] button

The listen button allows you to monitor the audio routed to the side-chain. This may be handy if you are processing this audio with the EQ or the filter (please read about it below).

It may be also quite handy to spot the exact frequency you want to trigger compression with, or to monitor external sidechain more accurately.

4.2.13.9. SC [Side-Chain] Processor Switch

This allows an extra control for the side-chain audio source. Options are Off (no extra control), EQ Peak band (the Presence band of the EQ), or Filter (the High-Pass section of the filter). This gives you extra control over the side-chain audio, to achieve a better Compressor gain-reduction behavior.

By default, this switch is Off.

4.2.13.10. SC Freq [Side-Chain EQ/Filter Frequency]

This control sets either the center frequency for the EQ Presence band (when you have opted for EQing the side-chain signal) or the filter frequency cutoff when the filter is the selected option.

Since this a double parameter, ranges and default values vary depending on the option. When the side-chain processing option is through the high-pass filter, the range for the frequency cutoff goes from 20 Hz to 2 kHz, and the the default frequency is 60 Hz.

If the side-chain processing option is the Presence EQ band, the frequency range goes from 20 Hz to 20 kHz, and the default frequency is 968 Hz.

4.2.13.11. SC [Side-Chain] Peak/Gain

This controls the amount of cutoff added to the side-chain signal by the High-Pass filter, or the amount of Gain for the presence band. Again, this is a double control parameter, and therefore, has different ranges and default values, according to the processing option.

If the side-chain processing option is the high-pass filter, the default is set to 0.800, and the the range goes from 0.100 to 2.00.

If the side-chain processing option is the Presence EQ band, the default is set to 0.00 dB, and the the range goes from -15.0 dB to 15.0 dB.

4.2.14. Saturation

The Saturation module adds saturation to the audio, as one would expect. Saturation is a subtle form of distortion that adds harmonics to the sound. This kind of distortion is related to the musical content, and therefore is perceived as musical and pleasant-sounding. The effect originates in the analog days, when audio was usually run through tube-based devices. The tubes distortion produces harmonics which contribute to enriching the sound, making it fuller, rounder and more colorful (what some would call 'warmth' or a 'warm sound', maybe because the tubes also produce heat). The harmonic saturation can also be found in magnetic tape recordings.

Our ears are very tolerant to harmonic distortion, and it has to be very loud to produce unpleasant results. That's why saturation is usually highly regarded and wanted.

You have four types of saturation, and controls for several parameters. A nice visualization in the Advanced Panel helps you better understand the effect the saturation will have on the audio.



4.2.14.1. Harmonics

This selects the kind of saturation you will obtain. Click the Thick button to be presented with 3 options: an emphasized (stronger) 2nd harmonic (the "Even" option), an emphasized (stronger) 3rd harmonic (the "Odd" option), or a third option (labeled "Thick") where both Even and Odd harmonics will be emphasized.

The Harmonics knob acts as a "drive" knob for the saturation. The more you increase the gain of this, the more amplified the saturation circuit becomes, leading to more saturation. The visualization screen reflects the amount of induced saturation by changing the waveform displayed. It goes from Off to 30 dB. Off is the default position (no saturation).

4.2.14.2. Overdrive

The Overdrive parameter is more about distortion than saturation. It works by pushing the waveform over the limit and squeezing it at that same limit, creating a flat line until the waveform comes back under it. Basically, it distorts the sound at the zero level, this way creating a high number of partials. The effect is way more drastic and coloring than the usual tube/tape saturation, hence the special control included for it in this section.

The visualizer clearly shows this kind of distortion applied to the waveform. However, the level of distortion is softened, to sound more pleasing. But you can still get a hard overdriving type distortion, if you combine Overdrive with one of the Saturation options.

The Overdrive control goes from Off (O) up to 50 dB. Off is the default value.

4.2.14.3. Make Up Gain

This is the Overdrive Output gain control. With this, you can compensate (either by attenuating or boosting, usually the first) the gain changes made to the audio by overdriving it. It is a volume control, and goes from -30 dB up to 12 dB. Default is 0.00 dB.

4.2.14.4. Curve

This control, when selected, increases the saturation induction curve by performing a "pre-emphasis" of the audio (think of it as a kind of "grit" parameter control). This "pre-emphasis" works by lowering the lows before the saturation stage. This way, the higher the saturation level, the more the mid and high frequencies are distorted relative to the low frequencies. This allows you to obtain the aforementioned 'grit' while preserving the definition, body and warmth given by the bass frequencies (which most of the time will simply be boosted, but not distorted).

After the saturation stage, there will be a "de-emphasis" process, to correct for the previous "pre-emphasis". This process works in the exact opposite way.

Curve is a toggle switch with only On and Off positions. By default it is On.



Since Curve will distort the high frequencies a bit more and the low frequencies a bit less, its effect depends very much on the type of audio being processed.

5. USER INTERFACE


The Graphical User Interface of the Bus FORCE follows the Arturia paradigm developed in the most recent effect plug-ins.

The GUI is built around a main control panel based on the looks of classical bus processing units. It also features an Upper Toolbar and a Lower Toolbar.

The Upper Toolbar contains the Preset Selection displaying the name of the currently active preset, preset selection filters and navigation arrows, and the very important Main Menu on the left (the 'hamburger' icon to the left of the plug-in name). It also has a button to access the Preset Library.

The Lower Toolbar displays the parameter names and brief explanations in the left-hand part. It also includes the Bypass button, a History button, Undo and Redo buttons, and a CPU consumption meter.

The parameter values appear in small floating window "cells" next to the parameter control. The values update in real-time when you move the controls.



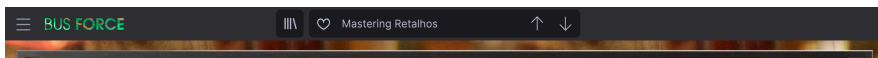
To know a parameter's current value, position the mouse over the corresponding control for about one second, and a small window appears next to it, revealing the value.

It is a very clear and straight-forward User Interface, and the visualization tools greatly expand its usefulness. That will always be the main focus of every Arturia product: to unleash your creativity while remaining easy to use.

We've already looked at the control panels. Now it's time to look at the toolbars.

5.1. The Upper Toolbar

The plug-in GUI (Graphical User Interface) has the usual Arturia toolbar that runs across the top, with the main menu / plug-in name on the left (the colored part), followed by the Library button and the Preset name, with arrows to navigate through the different presets stored in the library. The 'hamburger' button gives access to the Main Menu.



Let's now take a look at the Main Menu options. Since these options are also common to all current Arturia plug-ins, you may already be familiar with them:

5.1.1. New Preset

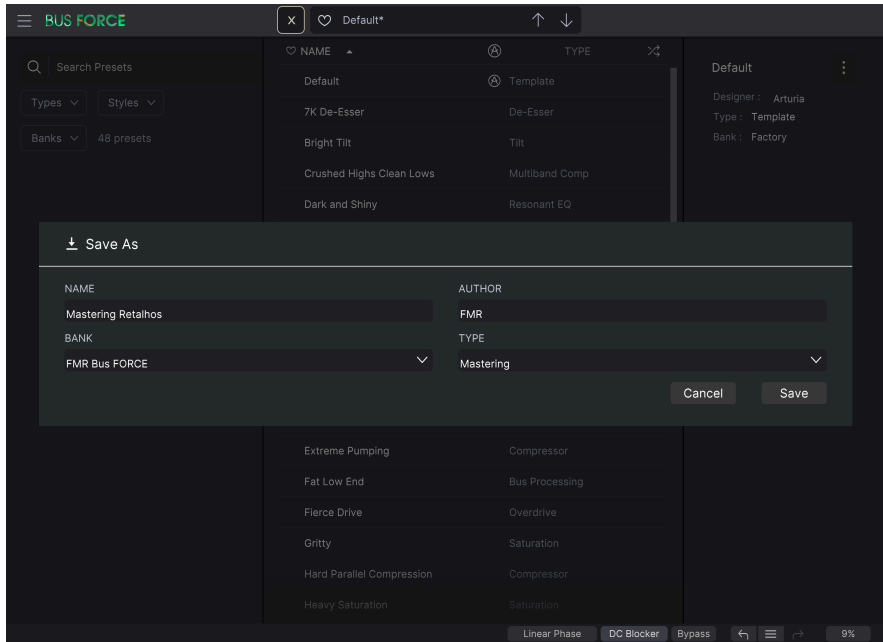
This option allows you to create a new preset based on the Default template.

5.1.2. Save Preset

This option will overwrite the active preset with any changes you have made, so if you want to keep the source preset too, use the Save As option instead. See the next section for information about this.

5.1.3. Save Preset As..

When you select this option, you are presented with a window where you can enter information about the preset. In addition to naming it, you can enter the Author name, and select a Type. You can even create your own Type by entering custom names in the Type field. This information can be read by the preset browser and is useful when searching for the preset later.



5.1.4. Import...

This command lets you import a preset file, which can be either a single preset or an entire bank of presets. Both types are stored in **.busx** format.

After selecting this option, the default path to these files will appear in the window, but you can navigate to whichever folder you are using to store presets.

5.1.5. Export Menu

You can export presets in two ways: as a single preset or as a bank.

- **Export Preset:** Exporting a single preset is handy when you want to share it with someone else. The default path to these files will appear in the "Save" window, but you can create a folder at another location if you like. The saved preset can be reloaded with the import preset menu option.
- **Export Bank:** This option can be used to export an entire bank of presets from the plug-in, which is useful for backing up or sharing presets.

5.1.6. Resize Window options

The Bus FORCE window can be resized from 50% to 200% of its original size without any visual artifacts. On a smaller screen such as a laptop, you might want to reduce the interface size so it doesn't dominate the display. On a larger screen or a second monitor, you can increase the size to get a better view of the controls. The controls work the same at any zoom level, but they can be harder to see at the smaller magnification values, or when using high resolution monitors (like HD monitors or higher). The higher the resolution, the bigger the size that should be used.

5.1.7. Tutorials

This option opens a panel on the right with Tutorials about the plug-in. Please try these to get more information and insights about the way this plug-in works.

5.1.8. Help

The Help section in this menu allows direct access to the User Manual (the document you are reading), as well as to the FAQ (Frequently Asked Questions).

5.1.9. About

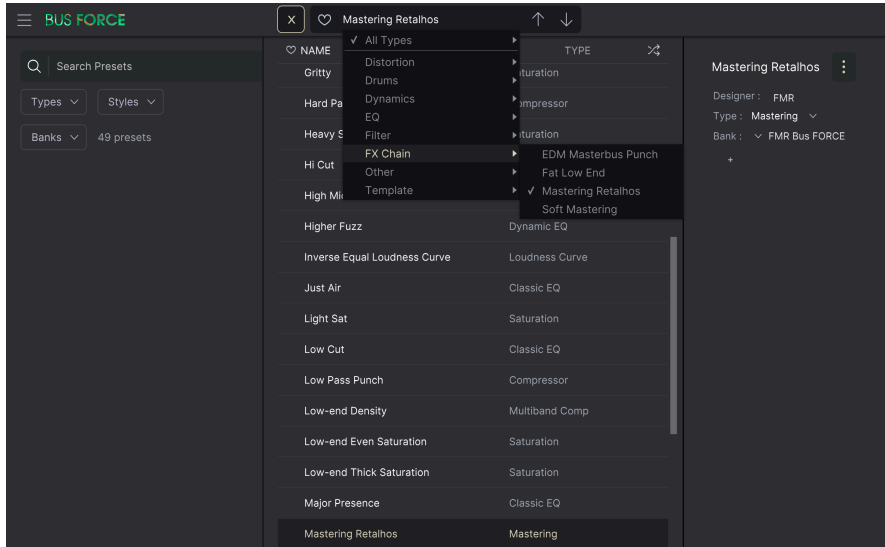
This opens the information window of the plug-in with the name, credits, and (more importantly) the number of the version installed. Don't forget to check frequently the Arturia Software Center for updates.

5.1.10. Preset Selection

The [Preset browser \[p.49\]](#) opens when you click the library symbol on the toolbar. The filter, name field and Up / Down arrows in the toolbar all assist with preset selection.

Select a preset by clicking the preset name field in the Upper Toolbar. This will open a list with all the presets available. The currently selected preset is marked with a ✓. Simply place the mouse over the name of the preset you want to select (the preset name will be highlighted), and click it.

Alternatively, you may use the Preset Up and Down arrows (the arrows to the right of the preset name field) to navigate through all the presets.

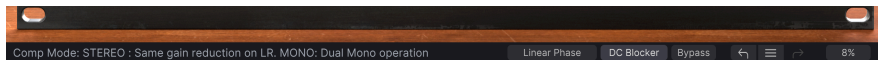


The Preset Selection list opens over the Preset Browser window. These are two complementary methods to look for and select presets.

5.2. The Lower Toolbar

When you hover the mouse over a parameter control, you will see a readout showing that parameter name and a brief description of it in the left part of the lower toolbar.

Also, you will notice that a small popup window appears next to the parameter control, displaying its current value. This will also show the value changes when you move the control (edit the parameter). This is handy, because you don't need to touch the parameter control to read the current value, and also you may keep looking at the parameter while you read the value changes.



On the right side of the lower toolbar are several small windows and buttons. These are very important features, so let's take a closer look.

5.2.1. Phase Response

This button is exclusive to the Bus FORCE and dedicated to mastering tasks. Linear Phase allows you to choose the phase response of the oversampling, selecting between linear phase or minimum phase.

When frequency bands are manipulated, their phase also reflects that manipulation. A phenomenon known as "phase shifting" may occur. When phases shift, they may cause "phase modulations" or even "phase cancellations" (frequencies oscillating out of phase change the overall volume, which may be perceived as a slow tremolo, or even as a modulation effect known as "phasing"). This happens because certain frequencies are being reduced, enhanced, or muted entirely depending on where the phased waves are colliding. The audio as a whole might present phasing problems because some frequencies might be missing while others are too pronounced.

This is the extreme scenario of "phase shifting". The issues are usually not that dramatic, and may be almost undetectable.

Bus FORCE always oversample. With the Linear Phase button, you may choose the behavior that best suits your purposes: linear phase at the expense of some latency, or minimum phase with no latency.

Linear Phase is great (and even advisable) when mastering, but due to the induced latency, it may not be so great (or advisable) when mixing.

The Linear Phase button is On by default, which is what you will probably want when using the Bus FORCE for mastering of bus duties.

5.2.2. DC Blocker

Although positioned in the Lower Toolbar, this button is exclusive to the Bus FORCE, so you won't find it in other Arturia plug-ins.

DC stands for Direct Current. DC blockers can perform ground loop elimination, signal source modulation leakage suppression, system signal-to-noise ratio improvement, etc.

Basically, when there is a deviation in DC, the audio oscillation center deviates to one of the sides, this way preventing the use of full digital scale resolution (one of the sides will reach peak much sooner than the other, because the oscillation center isn't at 0).

You don't need to turn on DC Blocker if you have proper recording sources, but even the modeled distortion can, in some settings, add a bit of DC.

As a matter of caution, you should always have a DC Blocker in a bus and mastering chain. So we have one in the Bus FORCE. By default, DC Blocker is On.

5.2.3. Bypass

This one is obvious. Activating the bypass option will completely disable the Bus FORCE plug-in. This action may also be performed by the Power switch. When the plug-in is bypassed, the GUI is grayed out and shows the word "Bypassed".

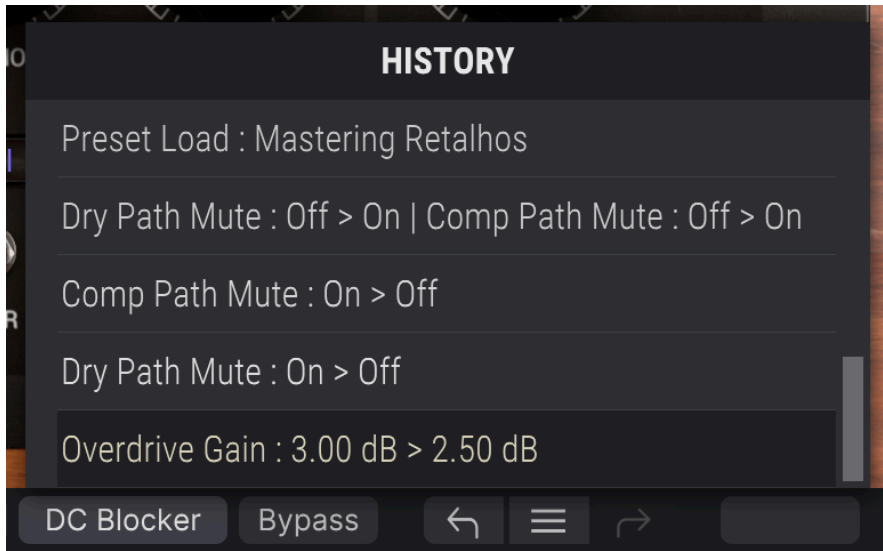


5.2.4. Undo

The Undo button is a curved arrow pointing to the left. This button reverses the last edit you performed. If clicked repeatedly, it will reverse the parameter changes in the order they were performed in the session, from the latest ones to the earliest ones.

5.2.5. History

This button lists all the latest parameter changes performed in the current session. You can always see the last four edited parameters, but using the mouse wheel or the navigation bar to the right allows you to access earlier edited parameters.



5.2.6. Redo

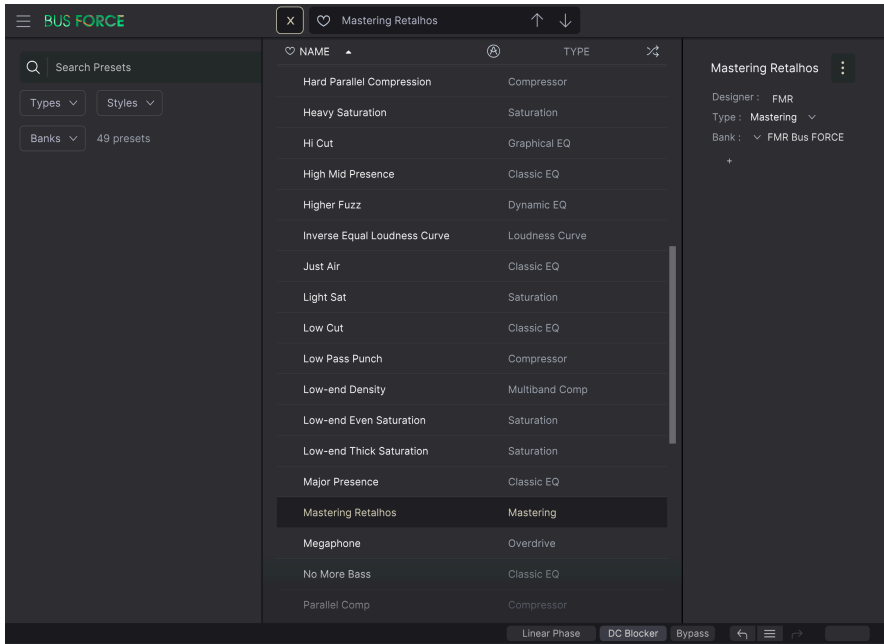
The Redo button is a curved arrow pointing to the right. This button works exactly the opposite way of the Undo button. It will restore the last undone edit. If clicked repeatedly, it will reinstate the parameter changes in the order they were undone (the latest ones first).

5.2.7. CPU meter

The CPU meter is used to monitor how much of your computer's CPU is being used by the plug-in. If you stress your computer too much, the global performance of your system and the audio may suffer.

5.3. The Preset browser

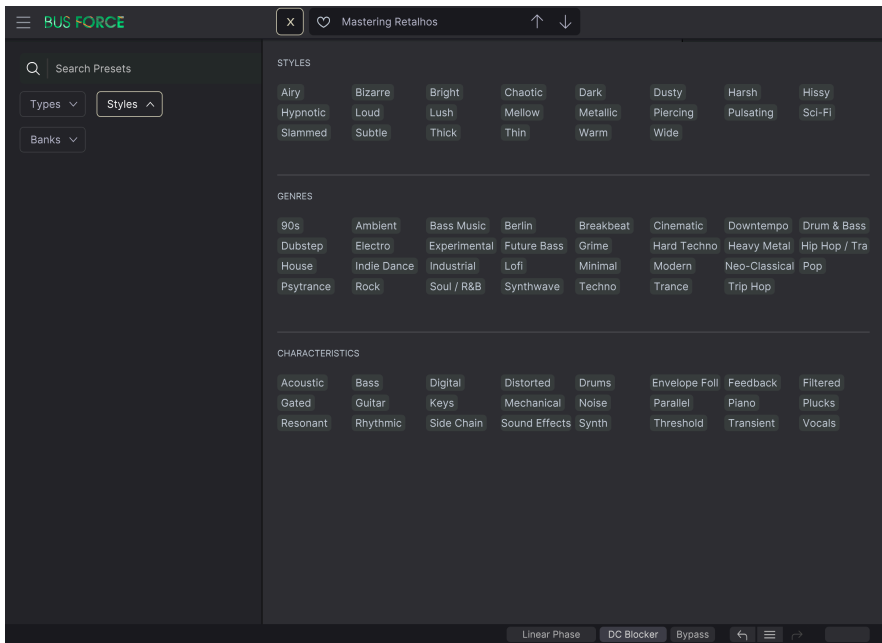
The preset browser enables you to search, load and manage preset configurations in Bus FORCE. Although this is based on the usual Arturia Preset Browser, it is simpler, and even easier to work with. You access the preset browser by clicking the library symbol next to the preset name field.



When you click the library symbol, you will see a screen with all the Presets you have saved. You can sort the list by several different criteria to make it easier to find the right preset. There are two columns. The first one can list the Presets by Name or by "Featured". The Featured presets are deemed of particular interest by Arturia, and are marked with the Arturia logo. The second one lists the Presets by Type.

There are other attributes you can use to filter the list. These are in the left column. There, you have the Types button, but also Styles, Banks, Genres, and Characteristics. If you click in one of these buttons, a list opens in the middle column, where you can select the tag you want to search for. A button appears in the left column, showing how many results there are for the tag you selected. Clicking that button will open the presets that fall under that classification. You may filter by more than one tag, in which case only the presets that fall under ALL the tags will be displayed. Beware that you don't filter too much or you might get no results.

You can also simply type a word in the search box (the one on the top left, to the right of the magnifying glass), in which case all presets whose name contains the word you typed in will be displayed. For example, let's say you want to search for a preset containing the word "drum". You simply type "drum", and all the presets containing those letters (even if the word is "drums" and not "drum") will be displayed. If the word is the name of a style, you may opt to choose by that style.

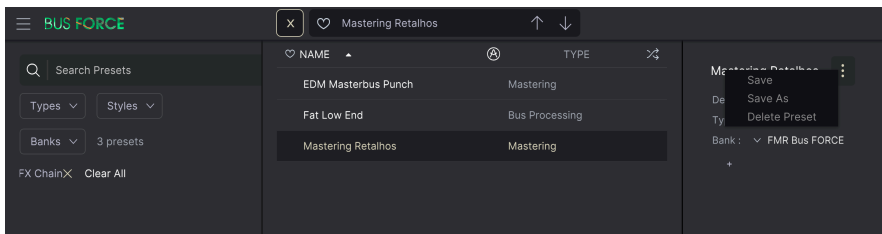


If you select Bank and choose a Bank, the Bank content will immediately be displayed.

When you click a preset name, that preset is selected, and you may see all the relevant information about it in the right column. There, you have the preset name, preset designer, type tag, bank it belongs to, and also all the other tags that have been added to it. If the preset belongs to a user bank, you also have a + sign, meaning you may add other tags. Tags are important, since they may help you quickly find what you are looking for, even among an extensive list of presets.

5.3.1. Delete Preset

If you want to delete a preset, first select it in the preset list. Next, click the library button to open the library. The Library window opens, with that preset still selected. In the right-hand column, you have the info for the preset, with the preset name. To the right of the name, you have a button with three dots. Clicking it opens a sub-menu with three options: **Save**, **Save As** and **Delete Preset**. Choose the 'Delete Preset' option. A dialog box opens, asking you to confirm. You may now confirm or cancel the operation. If you confirm, the preset will be deleted.



5.4. Fine-tuning parameters

Typically, to change values in the plug-in controls, just click the corresponding control and drag the mouse up or down. If the controls are switches, simply click them to toggle On or Off.

If you want finer editing values, you can use Ctrl+Drag (Cmd+Drag for macOS). Alternatively, you can Right-Click and Drag. With this technique the values change more slowly, which enables you to edit them with greater precision.

5.5. Resetting your controls

Double-clicking a control automatically changes it to the default value.

And that's it. We have just finished describing all the controls you have at your disposal to process sound in your DAW using the Bus FORCE plug-in. We hope you'll enjoy your new plug-in (and the results you get with it!) as much as we enjoyed making it.

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