Voxengo LF Max Punch User Guide

Version 1.16

https://www.voxengo.com/product/lfmaxpunch/

Contents

Introduction

Voxengo LF Max Punch is a professional audio effect plug-in for music and sounds where low-frequency thump and punch are most welcome, and where distortion is applied specifically to bring the bass sound to life. LF Max Punch provides a low-frequency effect specially designed for serious contemporary music producers, offering them a convenient tool for bringing a smooth punch and "oomph" to audio tracks and sounds.

LF Max Punch plug-in dynamically emphasizes selected bass-frequency band, applies a smooth saturation over it, and produces additional sub-harmonic content, with the ability to blend it with the original bass-band sound. Optional compression can then be applied to the resulting low-frequency sound. LF Max Punch first splits the broad-band signal into low- and high-frequency bands, and then applies the aforementioned effects to the lower band only.

This effect can be applied over a wide range of sound material: drum tracks, bass tracks, contemporary music mixes: rap, hip-hop, trance, club music; also rock music can benefit from LF Max Punch's low-frequency enhancements.

Features

- Puncher module
- Saturator module (3 modes)
- Sub-harmonic synthesizer
- Built-in compressor
- Effect monitoring switch
- Crossover steepness switch
- Stereo and multi-channel processing
- Internal channel routing
- Channel grouping
- Mid/side processing
- Up to 8x oversampling
- 64-bit floating point processing
- Preset manager
- Undo/redo history
- A/B comparisons
- Contextual hint messages
- All sample rates support
- Zero processing latency

Compatibility

This audio plug-in can be loaded into any audio host application that conforms to the AAX, AudioUnit, VST, or VST3 plug-in specification.

This plug-in is compatible with Windows (32- and 64-bit Windows XP, Vista, 7, 8, 10 and later versions, if not announced otherwise) and macOS (10.11 and later versions, if not announced otherwise, 64-bit Intel and Apple Silicon processor-based) computers (2.5 GHz dual-core or faster processor with at least 4 GB of system RAM required). A separate binary distribution file is available for each target computer platform and audio plug-in specification.

User Interface Elements

Note: All Voxengo plug-ins feature a highly consistent user interface. Most interface elements (buttons, labels) located at the top of the user interface are the same in all Voxengo plug-ins. For an in-depth description of these and other standard features, and user interface elements, please refer to the “Voxengo Primary User Guide”.

Crossover

Crossover stage splits the input audio signal in two spectral bands by means of the 24 dB/oct Linkwitz-Riley, 12 dB/oct or 6 dB/oct filter, depending on the crossover mode. The higher band is immediately sent to the output unprocessed. The lower band signal is sent as an input to various internal processing modules. However, if the “Puncher” module was enabled, the lower band signal is first processed by this “Puncher” module before going to other modules. Crossover parameters are specified for all channel groups to avoid phasing problems.

The “Freq” parameter specifies the corner frequency (in Hertz) of the crossover filter.

The “Punch” parameter selects the desired maximal dynamics change (in decibels) of the lower frequency band signal below the crossover frequency; note that this is not an equalizer gain: it is a dynamical change that affects transients only. This parameter will be available if the “Punch” switch was enabled (meaning the “Puncher” module is enabled). Depending on the sign of the value of this parameter the dynamics can be either increased or decreased.

Note that the crossover filter applies a relatively strong phase shift (coloration) over the frequencies around the corner frequency (up to 5.4 milliseconds of delay). In order to reduce changes made to the original tone of the important sounds in that spectral region you should fine-tune the “Freq” parameter precisely. Note that since the phase shift is non-linear, time-shifting the sound material you are processing with the plug-in may not always return phase coherence with other tracks.

Saturator

The “Saturator” module is used to produce saturated signal which can be mixed to the output. You may either use the full saturated signal bandwidth or use the supplied post filters to remove the high- and low-frequency content before mixing the signal to the output. Higher frequencies are best removed if you are using high “Drive” parameter values that produce a lot of higher harmonic content which usually sounds harsh. Lower frequencies can be additionally removed if you want to use the produced signal as a supplement to the original lower band signal: this technique can be used to fill the gap between the low and mid frequencies of the original signal, or to produce higher overtones that make the bass sound better on smaller speakers.

The “Saturator” module comes in two algorithms: the “Tubey” and “Legacy”, which has two types: the “Normal” and “Pumping”. The “Tubey” algorithm, which is quite loud in comparison to the “Legacy” algorithm, produces a warm tube-like saturation sound. The “Legacy/Normal” mode provides a kind of “linear” saturation: the higher the signal level the higher the distortion is. The “Legacy/Pumping” mode is more complex: it also reduces overall volume a little when the signal level is increased thus producing a “flowing” sound which allows you to avoid getting “oversaturated” bass-drum transients. Please note that output signal level in these algorithms and modes differs, making it necessary for you to make additional saturated signal gain adjustments when changing between algorithms and modes.

The “Drive” parameter specifies saturator’s drive amount in decibels.

The “Lo Cut” parameter specifies the corner frequency (in Hertz) of the high-pass filter (-6 dB/oct) that removes lower frequencies from the saturated signal.

The “Hi Cut” parameter specifies the corner frequency (in Hertz) of the low-pass filter (-12 dB/oct) that removes higher frequencies from the saturated signal.

LF Output Mix

This group of controls allows you to control the intermix of signals coming from different low-frequency (LF) signal processing modules of the plug-in, as they are sent to the low-frequency part of plug-in’s output signal.

The “LF Gain” parameter controls the gain of the input (“punched” or “non-punched” depending on the “Punch” switch’s state) low-frequency band in the LF mix. The “LF” switch can be used to remove this component completely from the LF mix.

The “Sat Gain” controls the gain of the saturated signal in the LF mix. The “Sat” switch can be used to disable the saturator module completely.

The “Sub Gain” controls the amount in decibels of synthesized sub-harmonic signal being added to the LF mix. The sub-harmonic synthesizer works with the input low-frequency signal. If the crossover filter is tuned to 128 Hz, this will generate a sub-harmonic signal that extends up to 64 Hz, one octave lower than the fundamental frequency. This works well in most cases and “thickens” the sound a lot, and even imparts a subtle presence effect. The “Sub” switch can be used to disable the sub-harmonic synthesizer completely.

The “Comp” switch activates the compressor stage. The compressor is applied over the LF mix before the final “Mix Gain” is applied. This is a feed-forward compressor with a fast attack and a medium release time. Compressor allows you to suppress excessive low-frequency peaks while still maintaining a good punch. In some cases, the punch effect can even be made stronger with this compressor.

The “Mix Gain” controls the overall gain of the LF mix before it is sent to the output.

The “Mono” switch enables mixing of the low-frequency signal mix to mono before it is sent to the output. This switch can be used as a quick “turn to mono” option for low frequencies that should not usually carry stereo information.

The “Monitor” switch enables monitoring of the low-frequency signal path alone. When this switch is enabled, you may adjust the “Dry Mix” parameter to evaluate the change your plug-in settings are doing to the original low-frequency sound. During monitoring, you may also toggle the “LF”, “Sat”, and “Sub” switches in various combinations, to hear how the corresponding signals mix with each other.

Out

The “DC” selector selects the DC filtering (DC offset removal) mode. Press the “Gears” button to open the “DC Filter Mode Editor” popup-window.

The “Dry Mix” parameter specifies the amount of dry signal being mixed to the output. This parameter allows you to fine-tune the presence of the processed signal in the final output.

The “Out Gain” parameter controls the overall output gain of the plug-in.

Credits

DSP algorithms, internal signal routing code, user interface layout by Aleksey Vaneev.

Graphics user interface code by Vladimir Stolypko. Graphics elements by Vladimir Stolypko and Scott Kane.

This plug-in is implemented in multi-platform C++ code form and uses “zlib” compression library (written by Jean-loup Gailly and Mark Adler), “base64” code by Jouni Malinen, FFT algorithm by Takuya Ooura, filter design equations by Magnus Jonsson and Robert Bristow-Johnson, VST plug-in technology by Steinberg, AudioUnit plug-in SDK by Apple, Inc., AAX plug-in SDK by Avid Technology, Inc., Intel IPP and run-time library by Intel Corporation (used under the corresponding licenses granted by these parties).

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Questions and Answers

Q. I'm trying to use LF Max Punch for an electric bass guitar. This bass guitar has an incredible tone that I'm trying not to change at all. All I simply want to do is create a sub-harmonic dimension to it. I can't seem to add this without slightly altering the essence of the tone. Do you have any settings suggestions to help me achieve this?

A. Make sure you have all modules but "LF" and "Sub" disabled, and use -6 or -12 dB/oct crossover mode. This way you can add sub-harmonics alone. However, sub-harmonics can alter the sound, and in some musical phrases sub-harmonics may sound dissonant. You may also try adjusting the corner frequency of the crossover filter to optimize the low-frequency timbre you get from the plug-in.

Q. It seems to work well but I'd love to have a limiting control so the bass-drum wouldn't clip.

A. You may use the compressor stage for the control over the low-frequency signal peaks.

Q. How is the saturation placed in the signal chain? Does it apply saturation across the entire LF band, or just to the region of interest (i.e. where you set the corner frequency)?

A. Saturation is applied to the whole LF band first, and then processed with the supplied cut filters.

Q. I don't understand the relation between X-Over frequency and the cut filters provided by the saturator module.

A. The "Lo Cut" and "Hi Cut" filters in saturator are applied after saturation takes place, to the saturated signal alone, and have no relation to the crossover frequency. Since strong saturation produces high-frequency content, it can be sometimes necessary to reduce it by means of the low-pass filter.

Q. Does the "Comp" compressor represent a one-knob compressor?

A. Indeed, it is a one-knob compressor like the one available in Crunchessor and Voxformer plug-ins. The parameter you adjust is a "drive" parameter. Negative "drive" values equate to a diminishingly smaller compressive action.

Q. When I'm using the Mid-Side mode, it seems that the "Monitor" switch picks the wrong channel.

A. If you would like to hear just the mid or side channel's low-frequency content, you have to additionally engage the "Solo" switch in the channel group selector.

Q. Do you think it's fine having LF Max Punch on a drum group channel, "punching" frequencies of both snare and bass drum?

A. LF Max Punch is very effective when used on the whole drum bus – it will simultaneously add punch to the bass drum and a bit of bite to the snare drum, especially if you engage the saturator.

Q. When using the -12 dB/oct filter, it sounds like LF Max Punch cancels out similar sounds on other tracks. Is it a bug?

A. This is not really a bug, it's a way -12 dB/oct filtering works – it reverses the phase above the corner frequency. If you experience such problem, please use -6 or -24 dB/oct filter instead.

Happy Mixing and Mastering!