KJAERHUS AUDIO

# MPL-1 Pro SE


High Precision Mastering Limiter

Second Edition

# Users Manual

## Copyright notice



## Trademark notice

# 1. Introduction

Thank you and congratulations on your choice of the MPL-1 Pro SE by Kjaerhus Audio

## About this Manual

This manual is made as a reference for working with the MPL-1 Pro SE - High Precision Mastering Limiter. It is written in a hierarchical manner.

### Symbol Explanation

To make it easy to find what you are looking for we are using a few symbols in the manual as seen below.

 Idea or proposal

 Note or info

## Acknowledgements


| | |
|---|---|
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| Manual: | Torben Larsen |
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| Beta Test: | Christopher Reis, Devon Brent, Jason Wolf, Lars Finnstrom, Marty Miller, Max Grønlund, Michael Kingston, Murray McDowall, Ted Perlman, Wolfgang Sperner, and several public beta testers at our support forum. |


## System Requirements

**Windows**

Windows 2000 / XP
500MHz Processor
256 MB RAM
VST 2.4 or RTAS 6/7 compatible host

**Mac**

OS X
500MHz Processor (PPC or Intel)
256MB RAM
VST 2.4, AU or RTAS 6/7 compatible host

## General Description

The Kjaerhus MPL-1 Pro SE is a limiter plug-in made for mastering and other purposes where the highest accuracy and quality are demanded. The MPL-1 Pro SE uses an over-sampled peak detecting algorithm to ensure that even high frequency peaks are accurately detected. This result in better high-frequency performance and it ensures that no overshoot or clipping occurs in the final D/A conversion. Look-ahead and a soft release-hold assist in keeping the compression smooth and transparent, even at high compression levels. The MPL-1 Pro SE uses a new stereo linking technique that allows peaks to be limited individually on each channel, while longer lasting gain reductions are linked. This technique ensures that peaks in one channel do not produce artifacts in the other channel, while the average compression still remains the same in each channel to avoid disturbances in the stereo balance. A PDR (Program Dependent Release) function is available to minimize pumping effect and increase loudness. The MPL-1 Pro SE also offers TPDF-Dithering for 8, 16 and 24 bit output resolution. Four stereo meters show the peak input and output levels as well as compression and RMS-output. Parameters can be controlled with remote devices through MIDI-Learn.


**Key Features**

- High-precision limiting with no overshoot
- Maximum loudness and minimum pumping
- Over-sampled peak detection with look-ahead
- Smooth Hold Function
- Stereo-link with unlinked peak limiting
- Four stereo meters with peak-hold
- AES17 compliant RMS output meter
- TPDF-Dithering for 8, 16 and 24 bit resolution
- A|B comparisons
- Silent knobs
- Low CPU usage
- Support sampling rates up to 96kS/s
- VST Automation
- Parameter control through MIDI-learn

# 2. Getting Started

In this chapter we will start by having a look at what is in the "package". Then we will discuss the reasons to use a mastering limiter and some of the important qualities of a good mastering limiter. We will also have a look of some of the techniques that make the MPL-1 Pro SE a "precision limiter".

## Plug-in Variants

The full installation of MPL-1 Pro SE will consist of VST/RTAS plug-ins for Windows and VST/RTAS/AU plug-ins for Mac (depending on the solution you bought). Each variant has different numbers of inputs and outputs and is intended for different applications as described below.

**VST**

MPL-1 Pro SE: Stereo version (2 in / 2 out).

MPL-1 Pro SE M: Mono version (1 in / 1 out).

**AU/RTAS**

MPL-1 Pro SE: Contains stereo, mono and mono to stereo versions (2/2, 1/1 and 1/2).

Banks and programs saved by one variant can be loaded into other variants as long as our proprietary file format (*.kja files) has been used. Programs or banks saved as VST presets (.fxp and .fxb files) can only be loaded by the variant they are saved with.

## Maximizing Audio

Limiting should be the last process in the mastering chain and is probably the most evident process indicating to the listener that the audio-material has been mastered. This is not because limiting makes the music sound much different, but because it makes it sound louder. However, making things louder is, as we see it, not a purpose of its own; there are better reasons to use a limiter.

**Good Reasons to use a Limiter**

Limiting is about using the dynamic area of the CDA, DVD, or any other destination medium optimally. If you analyze an un-mastered mix in a wave-editor you will probably find that it has a few peaks that are way over the average level of the song. A limiter can compress these peaks so the average level of the song can be increased. By utilizing the dynamic range of the destination medium properly, you increase the signal-to-noise ratio not only on the medium itself but also practically speaking on the listening system for playback.

With a limiter you can also make all the songs for an album sound equally loud (or whatever loudness relationship one would like) and still utilize the whole dynamic range of the medium on all songs.

There are other methods for taming peaks and increasing loudness, for instance using soft-clippers, tapes, etc, but these methods are absolutely not as transparent as using a good limiter.

### Never "Normalize" Audio

Some might be familiar with a function called "Normalize" found in many wave-editors. This function renders the audio-file to make the highest peak-level reach 0dB FS (Full Scale Digital). Normalizing audio is better than nothing, but it does not utilize the dynamic range of the destination medium as well as a limiter. Another problem is that even though all songs peak to 0dB FS they will probably not sound equally loud; this is because our ears detect signal levels based on the effective level (RMS) rather than peak-level. When you use a limiter you should never normalize as well.

### Use the Right Limiter

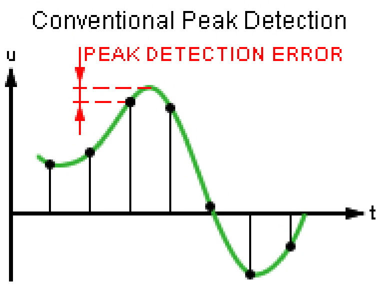
The main objective of a mastering limiter is to increase loudness as much as possible in a transparent way; meaning, we don't want the limiter to affect the audio-signal with too much sound of its own. As we are dealing with mixed material within the frequency range of 20 Hz to 20 kHz, it is important that the limiter works well in this frequency range without perceptible distortion, coloring or any other kind of artifacts, and it must be able to handle mixed material well. The MPL-1 Pro SE is especially designed for this purpose; other limiters may have other objectives.

## Precision Limiting

Precision limiting is like a meticulous surgical procedure, cutting exactly at the right spot without removing too much or too little. When dealing with digital signal processors some people might think that using a high word-length (bit-depth) on all calculations within the algorithms will automatically make everything exact, but this is far from the truth. Naturally, it is important to use a word-length high enough to make any rounding-errors insignificant in the final result, but there are far more troublesome error-sources to deal with; one of which is "peak detection errors".

### Peak Detection Errors

As the amount of compression in a limiter is based on the peak-levels of the audio signal, one of the most important objectives is to achieve accurate peak detection.



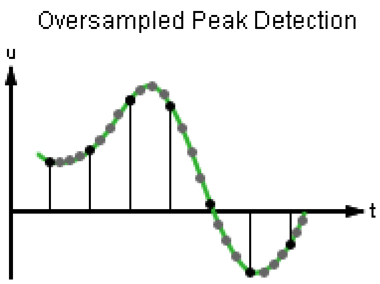

Conventional limiters compare the values of the samples in order to find the signal peaks on which the compression should be enacted. This might sound logical enough, but what if the "real peak" of the signal is somewhere in-between two samples (which would practically always be the case)? The result will be an error in our peak detection, as shown in the figure to the left, where the green line represents the originally sampled signal and the "pin needles" represents the samples.

Such peak detection errors are problematic in two ways. First, they result in artifacts that damage the clarity of the audio-signal in the high frequency region. Second, they will probably lead to digital distortion in the D/A converter of the audio-card or CD player. This last point might not be immediately apparent but the problem is that the peak detection error will cause the limiter to "under-compress" the signal, as it can not "see" how high the real peak-level is. Your meter will show that the output is peaking to 0dB FS, so it appears there is no overshoot occurring. However, in the D/A converter there is a digital reconstruction filter that will try to reconstruct the peak, but as the peak is above full range it will not be possible and the filter will clip (distort). In theory, peak detection errors can be as high as 3dB, making this a problem to take seriously, and the major reason why many mastering engineers run dynamic processors at over-sampling rates.

**Over-sampling Audio – A Poor Solution**

Over-sampling means that the audio-signal is converted to a higher sample-rate before the dynamics processor and then converted back to the original sample-rate afterwards. While this reduces peak detection errors, any sample rate conversion will unfortunately also add additional errors (noise) to our audio-signal, making it a poor solution.

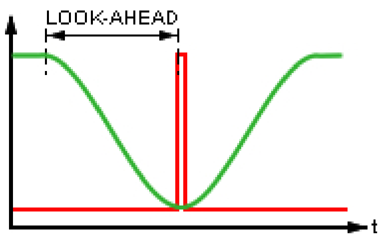
**The Best Solution**




The solution used in the MPL-1 Pro SE is to keep the sample rate of the audio-signal unchanged and only use over-sampling in the peak detector. This way, we have reduced the artifacts caused by peak detection errors by nearly 20dB without adding additional noise to the audio-signal. In addition, this solution saves a lot of CPU, as only a small part of the algorithm has to be run at a higher sample rate.

**Look-ahead Precision**

Another important precision factor is the look-ahead algorithm.




Look-ahead means that the limiter can "see" peaks a few milliseconds before they occur. This makes it possible to increase compression smoothly before the peak "arrives", which helps reduce compression artifacts. It is just like driving a car; as long as you can see what happens in front of you, you are able to drive smoothly and comfortably. In the figure to the left we see how the compression smoothly increases (green line) to make "room" for the upcoming peak (red line).

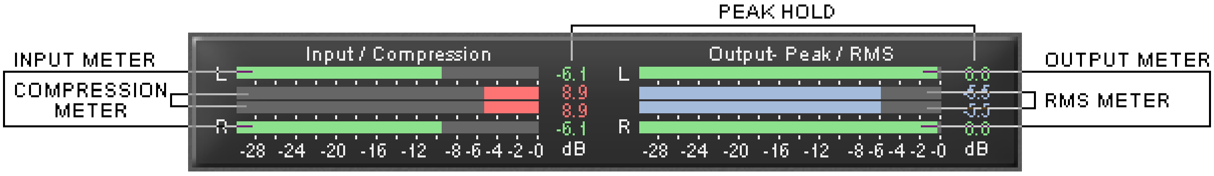
To obtain the best possible precision the MPL-1 Pro SE uses a special FIR filter to do the look-ahead compression "attack". The advantage of the FIR filter is that it has an exact settling time to the right amount of compression, which is not possible to obtain any other way. At the same time, the FIR filter is carefully designed to reduce audio-signal "leftovers" in the detector channel, which would otherwise lead to inter-modulation artifacts (side-bands).

# 3. Operating the Limiter

In this chapter we will discuss how to use the Limiter and add dithering.

## Meters

Four meters with peak-hold indicators are present in the meters section. These are used as reference when adjusting the parameters.




Input Meter: Shows the peak input level.

Compression Meter: Shows the peak compression.

Output Meter: Shows the peak output level.

RMS Meter: Shows the effective level. This meter can be used to evaluate the loudness of the output.

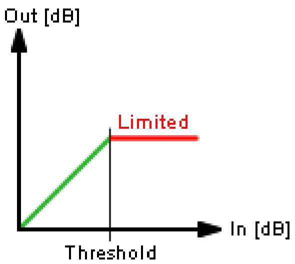
The RMS meter uses a first order integrator with a time-factor of 300 ms. It is compliant with AES17.

Peak Hold: Shows the maximum level or compression since last reset (or since GUI was opened) Right-click to reset.

## Basic Limiting

The limiter is mainly controlled by the six parameters seen below. In this section we will discuss the functionality of these parameters.


### Threshold




Threshold is the input level at which the limiter starts to compress the audio-signal. As the MPL-1 Pro SE has a hard knee, the limiter won't "touch" the audio-signal as long as its peak-level is below the threshold. If the peak-level is above the threshold the audio-signal will be ceiled to the threshold. Hard knee limiting produces the least amount of distortion for a given loudness, and is therefore preferred for mastering purposes.

If Auto-gain is activated the audio-level will increase when you lower the threshold.

**Release**

Release is the time it takes for the limiter to drop to a lower compression setting after the input signal level has dropped.

For most applications we recommend to set the release time between 100 and 500 ms and enable PDR (see below). If you are unsure about appropriate release times for your audio-signal, set the release time to 200 ms.

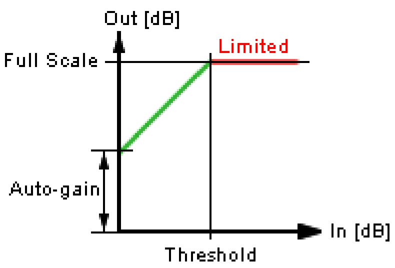
**Program Dependent Release (PDR)**

Setting release time has traditionally been a compromise between distortion, which is highest with a fast release, pumping, which is worst at medium release, and loudness, which is decreased the higher the release time is set. However, with the Program Dependent Release (PDR) function in MPL-1 Pro SE there is no longer any reason to compromise.

The PDR algorithm analyses the audio-material and makes adjustments to the release time dynamically. This way, loudness is increased while the compression remains free of distortion and pumping.

As opposed to an auto-release function the release time still remains active with PDR enabled, representing the highest possible release time. We recommend a release time between 100 and 500 ms.

**Auto-gain**




Limiting the audio-peaks can only make the output level go one way, down. We need to add additional gain to the output before we can harvest the benefit of the limiter.

The auto-gain function adds the necessary gain to ensure that the output level will peak to 0dB FS (Full Scale Digital) when the limiter is compressing. No more, no less.

**Gain**

The manual gain control can be used to adjust the output by up to +/- 5dB. When auto-gain is enabled it controls the output ceiling level.

**Stereo Link**

Stereo linking is used to ensure that both stereo channels are compressed equally. This is necessary in order to avoid disturbances in the stereo balance in cases where the signal level in one channel is louder, and therefore is compressed more than the other channel. Unfortunately, stereo linking also increases the total amount of artifacts produced when compressing; this is a physical law of stereo linking and therefore applies to all dynamic processors using conventional stereo linking.

Kjaerhus has solved this problem with a new patent-pending stereo linking system for the MPL-1 Pro SE. This stereo linking system allows peaks to be compressed unlinked while the effective level compression is linked. As our ears detect signal levels based on the effective level (RMS) rather than the peak-level, it is possible to maintain unlinked peak compression without disturbing the stereo balance as long as the RMS compression remains linked. As it is peak compression that produces the majority of artifacts, the new stereo linking system alone reduces artifacts by up to 6 dB compared to the conventional stereo linking method.

## Adding Dithering

Dithering should be used in all cases where we are going to reduce the word-length (bit-depth) of our audio-material; for instance when going from 24 or 32 bit audio to a standard 16-bit music CD. When we reduce the word-length of our audio material we are actually adding additional errors / noise to the material. A normal music CD has a signal-to-noise ratio close to 98 dB, which seems reasonably high compared to the signal-to-noise ratio of many analog systems, so why should we worry? The problem is that the sound of digital quantization noise is far nastier than the smoother sound of analog hiss. Dithering is noise with statistically distributed properties that, when added to the audio material before quantization, will literally turn the bad digital quantization noise into a more analog sounding hiss noise. It can also carry soft audio-signals through the quantization process that would have otherwise been too quiet to make it through on their own. The difference is most evident at very soft music levels (for instance when a song is fading out).


The MPL-1 Pro SE uses TPDF-Dithering (Triangular Probability Density Function) which is the “classic” dithering method. The signal level of this dithering is only +/-1 LSB peak-to-peak; lower than any other dithering. TPDF-Dithering does not alter the tonal balance of the audio-material and contrary to most noise-shaped dithering, TPDF-dithered material can be processed afterwards without getting an unpleasant surprise in the form of a loud noise floor in the processed material.

The available output bit resolutions are 8, 16 (CDA) and 24 (DVD).

Dithering can never make up for damage previously done to the audio, so if a signal has already been quantized to a smaller word-length it will not improve the sound quality to add dithering.

The dithering process pre-quantizes the audio so what you hear after Dithering is exactly the same as what you will hear after rendering the audio to a wave file of the same bit resolution as set in the dithering process.

## Expert Features

In order to get the most out of musical material, we have included a range of "expert features" that you won't see in many other limiters. With these features, you are able to make corrections to the way the limiter works. This is highly useful, as not all material is identical and therefore different music can benefit from different limiter behaviors.

### Input


If the input levels are out of range or badly balanced this can be corrected in the input section. This section is placed before any other sections or meters in the signal path of the MPL-1 Pro SE.

Gain: Gain adjustment (+/- 20dB), default 0dB.

Balance: Stereo balance (6dB law).

### PDR


The Program Dependent Release function decreases release time in dynamic moments of the music (for instance after a kick drum). This increases loudness and minimizes pumping without resulting in the high distortion that would occur with a permanently low release time.

Amount: How much the PDR function controls release time (0-100%), default 50%. At 0% PDR is off.

> Set this high enough to eliminate pumping and increase loudness. Increasing this further, after the pumping is gone and no significant increase in loudness is heard will just increase distortion.

Time: The time factor used to identify peaks in the music (40-400ms), default 126ms.

> Time and Amount are dependent parameters. If the time setting is low the amount would normally have to be rather high and vice versa.

### Compression


In this section look-ahead-time and hold-time can be adjusted. The hold function used in the MPL-1 Pro SE is improved so it instead of holding the compression at a certain level for a fixed amount of time, it allows a smooth transition between the hold state and release state. We have found this to sound far more transparent than the traditional release-hold. Information about look-ahead can be found in the "Getting Started" chapter.

Look-ahead: Look-ahead time (0-5ms), default 1.5ms.

Hold: Hold time (1-10ms), default 2ms.

### Stereo Link

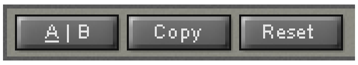

The unique, patent-pending stereo linking technique used in the MPL-1 Pro SE allows peaks to be compressed individually while the effective level compression (RMS) is linked. More information about this technique can be found under "Controlling the Limiter" in the "Getting Started" chapter.

Time: Time used to distinguish peaks from continuous signals (0-100ms), default 30ms. When set to 0ms it works like a conventional stereo link.

# 4. Parameter Handling

The MPL-1 Pro SE adds additional functions such as MIDI-learn (for parameter control), AB comparisons and Preset Filing. We will now explain these functions and how to operate controls on the user interface.

## AB comparisons



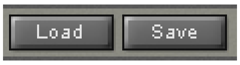

AB comparisons allow you to switch back and forth between two parameter sets to decide which sounds best.

A|B: Swaps between the two parameter sets.

Copy: Copies the current parameters (those you see now) to the other parameter set.

Reset: Loads the original factory program.

## Preset Filing




Programs can be saved to file. We use a proprietary file format with ".kja" extension.

**Loading a Factory Preset**

1. Click "Load" and a list of factory presets will appear.
2. Now click on the preset you want to load.

**Loading a Preset from file**

1. Click "Load" and a list of factory presets will appear.
2. Click "Load from file" at the bottom of the list and a file dialog will appear.
3. Select the file you want to open and press open in the file dialog.

**Saving a File**

1. Click "Save" and a file dialog will appear.
2. Now type in the name of the program or bank you are saving.

## Other Menu Options


This section contains the Manual, About and MIDI Setup buttons.

Manual: This button will open the manual you are reading right now.

About: This button will show a dialog with information about the plug-in name, version and license details.

MIDI Setup: This button switches between the menu section and the MIDI setup section.

# MIDI Setup

If you have a remote (hardware) controller that you want to use to control the parameters, it can be assigned through MIDI Learn. MIDI Learn is easy to use and only has to be set up once, as it is independent of programs and specific projects. All parameters can be assigned through MIDI Learn.


**Channel**

Channel determines the MIDI channel acknowledged by the plug-in. If set to “All”, the plug-in will respond to any MIDI channel (also known as Omni).

**MIDI CC Learn**

To assign a MIDI controller to a parameter, follow these steps:

1. Click the MIDI Learn button; the button will now begin to blink as long as it is enabled.
2. Operate the parameter on the user interface that you want to be able to control.
3. Operate a MIDI controller on your hardware controller; you will now see that the parameter begins to change on the user interface as you operate the controller.

Continue steps 2 & 3 for all the parameters you want to control through MIDI Learn. After you are finished, push the MIDI Learn button to turn it off.

If you did not succeed in assigning a controller to a parameter, it might be because the controller is already assigned to another parameter. In that case you need to un-assign the MIDI controller first (read below).

**MIDI CC Unlearn**

Un-assigning MIDI controllers are done following these steps:

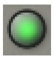
1. Click the MIDI Unlearn button; the button will now begin to blink.
2. Operate a MIDI controller on your hardware controller; it is now no longer assigned.

Repeat step 2 for all controllers that should be unassigned. After you are finished, push the MIDI Unlearn button to turn it off.

**MIDI CC Reset**

To reset all custom made MIDI controller assignments click the “Reset” button. After clicking “Yes” to the warning message that appears, the MIDI assignments will be back to the factory defaults.

## Operating Buttons



Standard-buttons toggle between off and on stages every time they are clicked.


Radio-buttons works like selectors where several knobs are grouped and only one can be active at a time. When one button is clicked the last active button goes off. In most groups one button stays active at all times but in special cases we made it possible to turn off all buttons.

## Operating Knobs


Knobs are operated by vertical mouse movement while holding down the left mouse button. Under the knob is a value-field that shows the parameter value. It is possible to change the sensitivity adjustment of the mouse and reset the knobs using interaction from the keyboard.

| | |
|---|---|
| Shift: | Holding down the Shift key makes the knob less sensitive to the mouse movements, which is useful for making fine adjustments. |
| Ctrl: | Holding down the Ctrl key while pushing the left mouse button resets the knob to its default value. |

Parameter values can be entered directly into the value field.

## Ghosted Controls

This plug-in comes in variants that share the same superset of controls. However, a balance or stereo link control does not make much sense in a mono version. To keep the same overall look between plug-in variants, and to make them compatible with each others presets, some controls are ghosted, as shown on the image below, to indicate that this particular parameter is ignored by the chosen plug-in variant.

# 5. Specifications

**Parameters**

| | |
|---|---|
| Threshold: | -30 – 0 dB FS |
| Release Time: | 5 – 5000 ms |
| Output Gain: | +/- 5 dB |
| Input Gain: | +/- 20 dB |
| Input Balance Pan-law: | Equal Level (6 dB) |
| PDR Amount: | 0 – 100% (at 0% PDR is off) |
| PDR Time: | 40 – 400 ms |
| Look-ahead Time: | 0 – 5 ms |
| Smooth Release Hold: | 1 – 10 ms |
| Stereo Link Time: | 0 – 100 ms |

**Dithering**

| | |
|---|---|
| Type: | TPDF (Triangular Probability Density Function) |
| Output Bit Resolution: | 8, 16 & 24 Bit |

**Meters**

| | |
|---|---|
| Input & Output: | Peak Sensing -30 – 0dB FS |
| Compression: | Peak Compression 0 – 30 dB |
| Output RMS: | AES17 (using a first order 300 ms integrator) |

All meters have a visual drop time of 15 dB per second. No peaks are spilled in the visual update.

**Other Data**

| | |
|---|---|
| Supported Sample Rates: | Up to 96k S/s |
| Latency: | 500 Samples |
| CPU Usage: | Less than 1.4 % on AMD Athlon® 64 3500+ |