

PSP MixPack v 1.5 Operation Manual

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Table of Contents

Table of Contents	1
End User License Agreement for PSP Software	3
PSP MixPack	4
Installation of PSP MixPack plug-ins on PC	5
PSP MixPack full version installation.....	5
PSP MixPack demo version installation	5
Uninstalling PSP MixPack from PC	5
Installation of PSP MixPack plug-ins on Mac	6
PSP MixPack full version installation.....	6
PSP MixPack demo version installation	6
Uninstalling from Mac	6
Support	8
A letter to the users of previous versions of PSP MixPack.....	9
PSP MixBass	10
General	10
Controls	11
Buttons	11
Knobs	11
Slider	12
Indicator	12
Meter	12
Settings.....	13
Structure and operation	14
Low frequency compressor	14
Bottom-end harmonics generator	14
Regulation of low frequencies.....	14
The system of soft-clipping signal peaks	14
Block diagram	15
PSP MixSaturator.....	16
General information	16
Controls	17
General controls	17
The analog saturation simulation algorithm.....	17
The bass frequencies processing algorithm.....	18
The treble frequencies processing algorithm	18
Peak and VU meters with 'overdrive' indicators	18
Settings.....	20
Structure and operation	21
The analog saturation simulation algorithm.....	21
The bass frequencies processing algorithm.....	24

The treble frequencies processing algorithm	24
Block diagram	25
PSP MixPressor.....	26
General information	26
Controls	27
Buttons	27
Knobs	27
Sliders.....	28
Peak level, VU and gain reduction meters with an overdrive signaling system	28
Settings.....	29
Structure and operation	31
Compressor.....	31
Limiter	33
Block diagram	34
PSP MixTreble	35
General information	35
Controls	36
Hiss Remover	36
Transients	36
Enhancer.....	37
Harmonics	37
Output.....	37
Settings.....	39
Hiss Remover	39
Transients section.....	39
Enhancer section	39
Harmonics section	40
Structure and operation	41
A general block-diagram of the MixTreble.....	41
Hiss Remover section.....	42
Enhancer section	43
Transients section.....	43
Harmonics section	45

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Piaseczno,
Poland.

PSP MixPack

PSP MixPack is a collection of VST plug-ins designed to improve the quality of digital sound recordings that often lack quality due to excessive digital sterility and harshness. As the name of the collection suggests, the plug-ins which make up part of this series are specifically for use during sound mixing. However, the quality of the algorithms employed also allows them to be used in mastering.

The kit consists of:

- » **PSP MixBass** a specialized low-frequency processor which is designed to assist in attaining a punchy, analog bass sound.,
- » **PSP MixSaturator** a specialized processor designed to produce a saturated sound, typical of such analog deices like tape recorders and valve circuits,
- » **PSP MixPressor** a compressor which enables the user to obtain sound quality typical of classic devices with valve and optoelectronic circuits,
- » and **PSP MixTreble** a universal high-frequency processor based on well-known analog devices.

Compatibility

PSP MixPack is compatible with applications which operate the standard VST plug-ins. The product has been tested in the following applications:

PC

- Cubase 3.6, 3.7, 5.x
- Nuendo 1.5
- Cubasis VST
- WaveLab 3.x
- Logic 4.x
- PARIS
- n-Track Studio
- Orion
- FXpansion VST-DX adapter 2 i 3

MAC

- Cubase 4.x i 5.x
- Cubasis VST
- Logic 4.x
- sonicWORX
- Spark

If your VST host application was not listed above we strongly recommend to install and test demo versions of our plug-ins for proper working with your VST environment. We would also appreciate if you informed us about your VST configuration so we could test it (contact@psp-audioware.com).

Limitations to demo versions

Demo versions of our-plug ins is the limitation in the signal processing. Demo versions turn off the signal processing every 15 secs for 1 sec. At the same time the plug-ins are switched to the mono mode.

Minimum system requirements:

- | | |
|------------------------------------|------------------------------------|
| • Windows 95 | • MacOS 8.5 |
| • 32 MB RAM | • 64 MB RAM |
| • Pentium 200 MHz | • G3 300 Mhz |
| • High Colour S-VGA 1024x768 | • High Colour S-VGA 1024x768 |
| • VST* compatible host application | • VST* compatible host application |

* The VST host application which supports the VST plug-ins operation affects directly the operation of the plug-ins from the PSP MixPack series and may limit its functional operation or even interfere with proper operation for which PSPaudioware.com s.c. accepts no responsibility.

You can use our VST plug-ins with DX compatible host applications using Expansion's VST-DX adapter.

Installation of PSP MixPack plug-ins on PC

PSP MixPack full version installation

To install the PSP's plug-ins on your PC please follow these steps:

- 1) run self-extracting archive, which will automatically start installation procedure,
- 2) enter your registration name
- 3) if you owned a common registration key for all PSP MixPack plug-ins check suitable check box next to the MixPack edit box, if you installed plug-ins using separate registration keys check check boxes next to plug-ins you wanted to install and then enter adequate registration keys to edit boxes,
- 4) if you also wanted to install mono versions of plug-ins please check the appropriate check box, mono versions will be installed with '_M' at the end of name,
- 5) select folder, where you want the operation manual to be installed and its language,
- 6) select folders, where you want plug-ins to be installed.*

PSP MixPack 1.5 plug-ins' files will be installed under names: PSP_MixBass_S.dll, PSP_MixSaturator_S.dll, PSP_MixPressor_S.dll, PSP_MixTreble_S.dll. Mono versions of plug-ins will be installed under names: PSP_MixBass_M.dll, PSP_MixSaturator_M.dll, PSP_MixPressor_M.dll, PSP_MixTreble_M.dll. Mono versions will be installed optional (after selecting suitable check box).

Monophonic versions are designed to use in applications which require one input - one output versions as mono insert effects, for instance Logic Audio (especially when combined with DSPF 2416). For technical reasons settings saved using one version are not usable with another one.

PSP MixPack demo version installation

- 1) run self-extracting archive, which will automatically start installation procedure,
- 2) choose plug-ins you want to install,
- 3) select folder, where you want the operation manual to be installed and its language,
- 4) select folders, where you want plug-ins to be installed.*

***Attention:**

The VST Cubase program keeps VST plug-ins in its own 'Vstplugins' subdirectory. For use with Cubase VST 3.72 or newer plug-ins can also be stored and run from another directory known as 'Shared VST Plug-ins Folder' common for many applications.

The LogicAudio program stores and runs VST plug-ins only from its own 'Vstplugins' subcatalog.

In order to install our product in other applications that accommodate VST plug-ins, the operating instructions of the relevant application should be referred to.

Uninstalling PSP MixPack from PC

- 1) Open the "MixPackInstall.txt" or "MixPackDemoInstall.txt" file available from the Start|Programs|PSPaudioware.com menu which contains all the information about installed components
- 2) If you want to uninstall full versions of the plug-ins, you should also remove the registry entries listed in the "MixPackInstall.txt" file using the regedit.exe program
- 3) Remove the operation manual as given in the "MixPackInstall.txt" or "MixPackDemoInstall.txt" file
- 4) Remove all plug-in copies listed in the "MixPackInstall.txt" or "MixPackDemoInstall.txt" file

Installation of PSP MixPack plug-ins on Mac

PSP MixPack full version installation

To install the PSP's plug-ins on your Mac please follow these steps:

- 1) extract files from SIT archive,
- 2) move VSTFx plug-in files with ‘_S’ and the end of name to suitable ‘Vstplugins’ folder*,
- 3) if you wanted to use mono version of MixPack plug-ins you should also move file with ‘_M’ at the end of name,
- 4) move documentation files where you want to keep them,
- 5) if you owned a common registration key for all PSP MixPack plug-ins run ‘PSP_MixPack_Registration’ application, in other case use separate registration applications for individual plug-ins: PSP_MixBass_Registration, PSP_MixSaturator_Registration, PSP_MixPressor_Registration and PSP_MixTreble_Registration,
- 6) enter your ‘registration name’ and press ‘return’ key,
- 7) enter your ‘registration key’ and press ‘return’ key.

PSP MixPack 1.5 plug-ins’ files will be installed as VSTFx files: PSP_MixBass_S, PSP_MixSaturator_S, PSP_MixPressor_S, PSP_MixTreble_S. If you also wanted to install mono versions please copy PSP_MixBass_M, PSP_MixSaturator_M, PSP_MixPressor_M, PSP_MixTreble_M files to adequate ‘Vstplugins’ folder

Monophonic versions are designed to be used in applications which require single input and output plug-ins, as mono insert effects, for instance Logic Audio (especially if it cooperates with DSPF 2416). For technical reasons, settings saved using one version of the plug-in are not usable with another one.

The PSP MixPack 1.5, unlike previous versions, contains additional mono versions of the plug-ins: PSP_MixBass_M.dll, PSP_MixSaturator_M.dll, PSP_MixPressor_M.dll, and PSP_MixTreble_M.dll for PC, and PSP_MixBass_M, PSP_MixSaturator_M, PSP_MixPressor_M, and PSP_MixTreble_M for Mac. They are installed optionally (after choosing the required box in the installation program on PC or by simple move operation on Mac) to be used in host applications which require single-input and output plug-ins, as insert effects for mono tracks, for example, Logic Audio (especially if it cooperates with the DSPF 2416 card). For technical reasons, settings saved using one version of the plug-in are not recognized by the other version.

PSP MixPack demo version installation

- 1) extract files from SIT archive,
- 2) move VSTFx plug-in files to adequate ‘Vstplugins’ folder,
- 3) move documentation files to folder where you want to keep them.

***Attention:**

The VST Cubase program keeps VST plug-ins in its own ‘Vstplugins’ subdirectory. For use with Cubase VST 5.0 and newer plug-ins can also be stored and run from another directory known as ‘Shared VST Plug-ins Folder’ common for many applications.

The LogicAudio program stores and runs VST plug-ins only from its own ‘Vstplugins’ subcatalog.

In order to install our product in other applications that accommodate VST plug-ins, the operating instructions of the relevant application should be referred to.

Uninstalling from Mac

- 1) If you want to uninstall full versions of the plug-ins, you should remove the “PSPmb.reg”, “PSPms.reg”, “PSPmp.reg”, and “PSPmt.reg” files from the “Preferences” system folder
- 2) Remove plug-in documentation files

- 3) Remove all plug-in copies: PSP_MixBass_S, PSP_MixBass_M, PSP_MixSaturator_S, PSP_MixSaturator_M, PSP_MixPressor_S, PSP_MixPressor_M, PSP_MixTreble_S, and PSP_MixTreble_M.

Support

If you have any questions or doubts as to the principles and operation of our plug-ins, please visit our website www.pspaudioware.com you can find there the latest product information, free software updates and answers for frequently asked questions.

You can also contact us using e-mail: support@psp-audioware.com. We will gladly answer your questions. As a rule we respond within 24 hours.

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A letter to the users of previous versions of PSP MixPack

If you are a user of earlier versions of the PSP MixPack plug-ins, the main change in the installation process is the use of the same installation program for all the plug-ins regardless of whether a single plug-in or the whole series is being installed.

The 1.5-version files of the plug-ins will be installed under the following names on your PC: PSP_MixBass_S.dll, PSP_MixSaturator_S.dll, PSP_MixPressor_S.dll, PSP_MixTreble_S.dll, or as VSTFx files: PSP_MixBass_S, PSP_MixSaturator_S, PSP_MixPressor_S, and PSP_MixTreble_S on Mac, due to which fact you still will be able to use the previous versions of our plug-ins.

The PSP MixPack 1.5, unlike previous versions, contains additional mono versions of the plug-ins: PSP_MixBass_M.dll, PSP_MixSaturator_M.dll, PSP_MixPressor_M.dll, and PSP_MixTreble_M.dll for PC, and PSP_MixBass_M, PSP_MixSaturator_M, PSP_MixPressor_M, and PSP_MixTreble_M for Mac. They are installed optionally (after choosing the required box in the installation program on PC or by simple move operation on Mac) to be used in host applications which require single-input and output plug-ins, as insert effects for mono tracks, for example, Logic Audio (especially if it cooperates with the DSPF 2416 card). For technical reasons, settings saved using one version of the plug-in are not recognized by the other version.

Attention: In some VST host applications, when you choose the mono-version option of the installation of our plug-ins, problems with previous versions of the **PSP MixPack** plug-ins may occur. In order to prevent those problems, we recommend the manual removal of the old versions of the plug-ins files: PSP_MixBass.dll, PSP_MixSaturator.dll, PSP_MixPressor.dll, and PSP_MixTreble.dll on PC, and VSTFx files: PSP_MixBass, PSP_MixSaturator, PSP_MixPressor, and PSP_MixTreble on Mac.

Because of the changes in the internal structure of the 1.5-version plug-ins, you will not be able to use the settings which were saved in earlier versions of the **PSP MixPack** with the new versions. The new versions of the plug-ins will not be recognized in the projects which used their old versions.

Main changes to previous versions:

- » separate mono versions for host applications which require the mono-mono mode of operation have been introduced, which enable, among other things, the comfortable application of the plug-ins in Logic Audio, especially when the DSPF 2416 card is used.
- » the Mix slider enables a smooth change in proportion between the input signal and processed signal, but before the output section of the plug-in
- » improved graphic operations for better refreshing of the image and correct display of controls
- » correct display of values between -1. and 0.
- » in most host applications the knobs operate in a linear mode; in Cubase 5.0 and Nuendo 1.5 host applications the knob operation is set by the superior host application
- » some minor errors have been removed in order to increase the plug-ins compatibility and overall reliability
- » indicators of the peak meters in PSP MixSaturator and PSP MixPressor are held for 300ms as opposed to 1000ms in previous versions of these plug-ins
- » basic plug-in parameters can be correctly automatized in Cubase
- » the library of presets for PSP MixSaturator and PSP MixPressor has been improved and extended.

PSP MixBass

General

PSP MixBass is a specialised low frequency processor which is designed to assist in attaining a punchy, analog bass sound. In order to ensure a wide range of sound capabilities while retaining a high sound quality, a series of innovative solutions are employed with this plug-in:

- » a specially designed low-frequency compression algorithm which enables the user to adapt dynamic bass features to the psycho-acoustic properties of the ear,
- » a bottom-end harmonics generator which adds character and definition to the range of bottom end frequencies, particularly in relation to synthetic bass and percussion loops,
- » a soft (pseudo-analog) clipping algorithm which prevents the occurrence of digital distortion when exceeding 0dBfs as well as allowing an increase of volume to around 3dB without a deterioration in the quality of recording.

Features

- » a wide range of possible sounds,
- » psycho-acoustic compression of low frequencies,
- » additional boosting of bottom-end frequencies through the addition of harmonics,
- » the soft clipping algorithm in the event of exceeding 0dBfs,
- » mixing of input signal with processed one in the desired proportions,
- » contains a set of factory presets.

Applications

- » for the enrichment of the low-frequency sound during recording, mixing and mastering,
- » mix processing and single-track recordings (percussion loops, synthetic bass and other acoustic, electric and electronic instruments).

Specifications

Plug-in type: Plug-in VST, PC and MacOS version

Internal signal processing: 32-bit, floating-point

Internal latency: 256 samples, compensated

Processing speed: below 10% of the computation power of the PII 333MHz or G3 300MHz

Controls

The operation of the parameter settings of the PSP MixBass plug-in has been standardized to be compatible with other elements of the VST system.

- » Control settings are changed by clicking or clicking and dragging.
- » Clicking and dragging the slider while holding down the 'shift' key ensures precision in changing the slider settings.
- » Knobs operate in a linear mode by default, however you could press 'alt' key to operate in circular mode; in Cubase 5.0 and Nuendo 1.5 host applications the knob operation is set by the superior host application.
- » Clicking on a knob or a slider while holding down the 'Ctrl' key will restore the standard setting.
- » Clicking on the name of the plug-in causes the window 'information about' to appear.



Buttons

[process]

A button which turns the compression on and off and adds low-frequency harmonics. It does not affect the operation of the 'tune', 'bass' and 'gain' knobs or the 'saturation' button.

[oo/o]

Stereo/mono switch. It is set automatically when the plug-in is turned on but manual setting is also possible.

[saturation]

This turns on the system of soft-clipping signal peaks when sound levels are too high. It enables the user to reduce digital distortion when exceeding 0dBfs.

Knobs

[tune]

Regulation of the frequency of the filter which divides the input signal into low range frequencies (which undergo further processing) and the remaining frequencies.

[threshold]

Regulation of the compressor threshold. Unlike classic compressors, the algorithm used here compresses low frequency signals at a level beneath the threshold which is set, with a ratio of 2:1.

[colour]

Regulation of the boosting of the bottom-end frequencies, together with the addition of the harmonics (both odd and even) to this range.

[bass]

Regulation of the content of low frequencies in the output signal.

[gain]

Regulation of the level of the output signal (before saturation)

Slider**[mix]**

Regulation of the input to processed signal ratio. It allows to set processing depth of plug-ins' algorithms precisely.

Indicator**[peak indicator]**

Indicates the possibility of signal overdrive (reaching or exceeding the threshold of 0dBfs for longer than three samples). Overdrive is indicated considerably less when the soft clipping system is on ('saturation') than when it is off.

Meter**[gain meter]**

Indicates how much (0..+48dB) the low frequency level has been strengthened as a result of compression. The higher the threshold ('threshold'), the higher the level on the meter (the indicator moves further to the right).

Settings

The sound obtained with the **PSP MixBass** is dependant on the type of phonic material being processed. There is no universal recipe for achieving satisfactory effects. Production presets are only initial propositions which serve as a departure point for further experimentation.

PSP MixBass is excellent for processing percussion loops. Therefore, any of the possible settings within the plug-in parameters can produce interesting effects.

When working on individual bass tracks, the settings of **PSP MixBass** should be adapted to the particular bass instrument. For example, with the acoustic bass-guitar the addition of a large quantity of harmonics by using the 'colour' parameter will cause a loss of the instrument's natural sound. Alternatively, with a uniform synthetic bass sound it is even possible to use the maximum setting of the parameter.

When using **PSP MixBass** for mastering it is advisable to use the settings which delicately correct the sound recording (the 'mastering' presets) and to carefully monitor the quality of the output signal on the best available monitoring system.

Using 'Mix' slider allows you to set optimum processing depth without a need to change controls' settings of individual algorithms. It also makes possible mixing clean input signal with deeply processed and distorted signal in any proportion to obtain special effects.

The **PSP MixBass** has been equipped with a library of 21 presets. These enable the user to familiarise themselves with the plug-in's possibilities in relation to the processing of different signals, and also serve as a quick guide to particular algorithms and their parameters. To make things easier, the presets have been divided into eight groups:

Group	Application
<i>psychoacoustic</i>	adaptation of the signal to psychoacoustic features of hearing
<i>mastering</i>	mastering process
<i>warm</i>	the warming of the bass range
<i>drum-loop</i>	drum loops
<i>bass</i>	bass instruments
<i>guitar</i>	guitars and other instruments
<i>disco</i>	special bass frequency overdrive
<i>extreme</i>	extreme distortion

Structure and operation

Low frequency compressor

The low frequency compressor ('threshold' control) facilitates the definition, softening and lengthening of the bass sound as well as picking out the less audible articulatory details and sounds in the background.

Unlike compression systems in general use, the algorithm used in **PSP MixBass** processes signals whose level comes below the chosen threshold (which is regulated by means of the 'threshold' control). The cut-off frequency of the filter is regulated by the 'freq' control. The compression ratio is 2:1 and corresponds to the narrowed range of audible dynamics in the low frequency range.

The positive effects of applying this kind of compression are that it gives depth to the processed material, strengthens the coherence of middle range and low range frequencies and provides a stable base for the bass in music.

Inappropriate use of this compression may cause blurring and making a mass of the bass through which the whole recording may lose its definition and dynamics.

Bottom-end harmonics generator

The bottom-end harmonics generator ('colour' control) is used for achieving a clear and characteristic bass contour. By using this function you can give the recording a unique character, for example, of a saturated, overdriven analog tape.

These aims are realised through the boosting of the bottom-end frequencies as well as their enrichment in odd and even harmonics. The cut-off frequency of the filter is dependant on the setting of the 'freq' control in such a way that the harmonics obtained through the operation of the system are situated within the range set.

Regulation of low frequencies

This function ('bass' control) enables the user to regulate the content of low frequencies, similar to the shelving equaliser with its regulation of the cut-off frequency of the filter. Due to the characteristics of the filters, extremely fine regulation of the recording sound is possible.

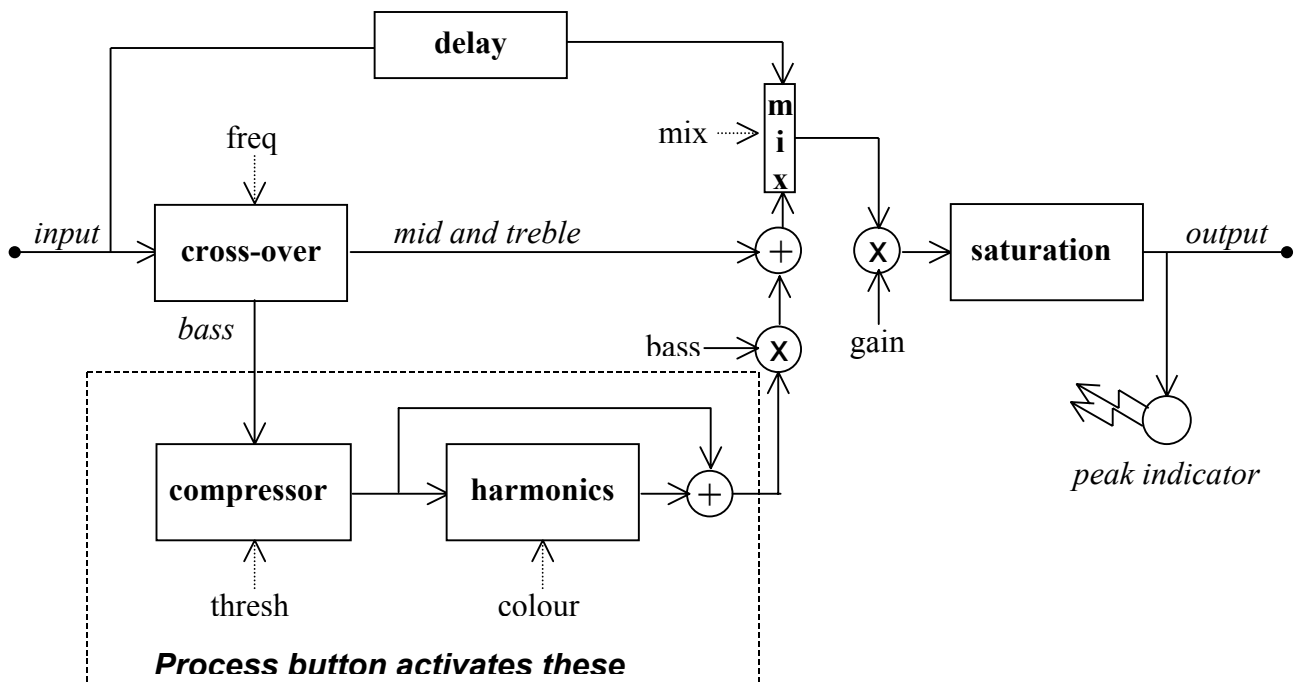
The system of soft-clipping signal peaks

This function is designed to reduce digital overdrive in a signal when exceeding 0dBfs, by using soft pseudo-analog saturation. ('saturation' control)

This system has a positive effect on the sound of high-amplitude signals and facilitates an increase in the overall recording level to around 3dB without audible distortion. This function can also be used to obtain the effect of deep overdrive.

Like other plug-ins from the MixPack 1.5 series, the MixBass 'mix' control can be used for mixing the input signal with the processed signal in the desired proportions. This function allows for quick and comfortable setting of the optimum depth in almost all plug-in algorithms. This regulation is before output section of the plug-in and does not influence directly to the operation of soft clipping algorithm.

Block diagram



Specific description of the block diagram for the PSP MixBass processor

<i>input</i>	plug-in input
<i>output</i>	plug-in output
<i>cross-over</i>	a set of filters separating low and medium to high frequencies
<i>freq</i>	cut-off frequency of filters
<i>mid and treble</i>	medium and high frequencies
<i>bass</i>	low frequencies
<i>compressor</i>	low frequency compressor
<i>thresh</i>	threshold of the compression
<i>harmonics</i>	bottom-end harmonics generator
<i>colour</i>	regulation of the harmonics content
<i>bass</i>	regulation of the low frequency content
<i>gain</i>	regulation of the strength of the output signal
<i>mix</i>	regulation of the proportion of input signal to processed one
<i>saturation</i>	system of soft-clipping signal peaks
<i>peak indicator</i>	signal of overdrive output
<i>process button activates these sections</i>	processor elements within the frame are active if the 'process' button is on.

PSP MixSaturator

General information

PSP MixSaturator is a specialised processor whose aim is to produce a saturated sound, typical for such analog devices as tape recorders and valve circuits. In the interests of ensuring a wide range of sound possibilities and high sound quality, three algorithms have been used in this plug-in:

- » the analog saturation simulation algorithm, which enables the user to choose one of the seven curves of non-linearity characteristic of valve devices, analog tape and digital clipping,
- » the bass frequencies processing algorithm, whose aim is to add warmth to the sound within the bass frequencies range, through adding harmonics and increasing the bass in a recording,
- » the treble frequencies processing algorithm, which gives the user the possibility of simulating tape saturation within the treble frequencies range, without increasing the level of distortion and aliasing.

Features

- » the simulation of saturated sound characteristic of valve devices and tape recorders,
- » a wide range of possible sounds,
- » special circuits to shape the sound of bass and treble frequencies,
- » mixing of input signal with processed one in the desired proportions,
- » contains a set of factory presets.

Applications

- » the enriching of sound with harmonics and increasing the mean level while recording, mixing and mastering,
- » the processing of mixes and single tracks.

Specifications

Plug-in type: Plug-in VST, PC and Mac OS version

Internal signal processing: 32-bit, floating-point

Processing speed: below 14% of the computation power of the PII 333MHz or G3 300 MHz processor with all algorithms set on

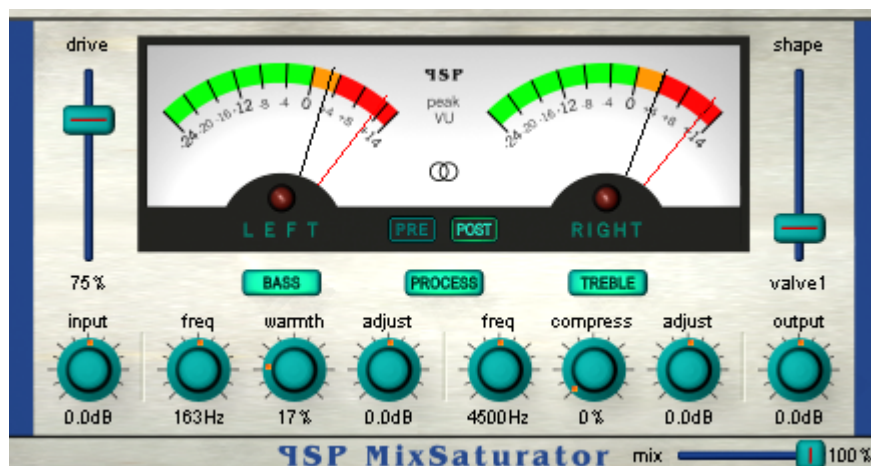
Szybkość przetwarzania: poniżej 14% zajętości procesora PII 333 MHz lub G3 300 MHz

Internal latency: 0 samples

Controls

The operation of the parameter settings of the PSP MixSaturator plug-in has been standardized to be compatible with other elements of the VST system.

- » Control settings are changed by clicking or clicking and dragging.
- » Clicking and dragging the slider while holding down the 'shift' key ensures precision in changing the slider settings.
- » Knobs operate in a linear mode by default, however you could press 'alt' key to operate in circular mode; in Cubase 5.0 and Nuendo 1.5 host applications the knob operation is set by the superior host application.
- » Clicking on a knob or a slider while holding down the 'Ctrl' key will restore the standard setting.
- » Clicking on the name of the plug-in causes the window 'information about' to appear.



General controls

[process] button

Turns the processing in the plug-in on and off. It does not affect the "input" knob.

[oo/o] button

Stereo/mono switch, which is set automatically during the initialisation of the plug-in. It can also be set manually if necessary.

[input] knob

Regulates the input signal level. It is active independent of the "process" button

[output] knob

Regulates the output signal level. It is active only when the process button is lit up.

[mix] slider

Regulation of the input to processed signal ratio. It allows to set processing depth of all plug-ins' algorithms precisely.

The analog saturation simulation algorithm

Every signal, which after initial processing with bass and treble frequencies algorithms and before employing the "drive" slider has a value of 0dBFS, will have the same value on exiting the algorithm. This is why, in most cases, if high values for the processing parameters of bass frequencies have not been set, the output signal should not exceed the maximum level. If the changes caused by the work of the bass algorithm are significant enough that there is a need to change the output level, the "output" button should be used. Any signal exceeding 0dBFS on exiting the algorithm is indicated by an LED diode in the "post" setting of the meter.

[drive] slider

The "drive" control of the main analog saturation simulation algorithm. The operation of this slider depends on the selected nonlinearity characteristics.

[shape] slider

The shape slider is for choosing the shape of the nonlinearity curve of the analog saturation simulation algorithm. There are six available characteristics for analog devices, as well as a curve typical for digital clipping (slider up) and a setting which skips the main analog saturation simulation algorithm (slider down). In order to find details on these characteristics, go to the description of the curves.

The bass frequencies processing algorithm**[bass] button**

Turns on the processing of bass frequencies. It activates the knobs below this button which have an influence on the processed signal.

[freq] knob

Regulates the bass frequencies filter cut-off.

[warmth] knob

Regulates the bass content with its warm harmonics.

[adjust] knob

Regulates the bass frequencies level.

The treble frequencies processing algorithm**[treble] button**

Turns on the processing of bass frequencies. It activates the knobs below this button which have an influence on the processed signal.

[freq] knob

Regulates the treble frequencies filter cut-off.

[compress] knob

Regulates the depth of compression of the treble frequencies.

[adjust] knob

Regulates the treble level.

Peak and VU meters with 'overdrive' indicators

The **PSP MixSaturator** meters are complex measuring devices, which are used to establish the work-levels of the algorithms in the plug-in, the levels of the output signal and the dynamic accuracy of the processed signal.

The meters have been given a logarithmic scale. The difference between the peak value and VU for typical music signals is 14dB, hence the maximum value on the scale is +14 which is equal to 0dbBFS, with the value 0 being equal to the VU level of a correctly made recording.

red indicator

This indicator shows the peak level of signals, which is measured with an accuracy to one sample. The maximum level is held for 300ms.

black indicator

This indicator shows the average-level VU of the signal. The integration time is 300ms. The indicator shows a level equal to the peak signal for the established sinusoidal signal.

overdrive indicators

LED diodes light up on reaching or exceeding +14 on the scale (0dBFS).

[pre/post] switch

Sets the meters indications mode:

- **pre** - the meters show levels after the level of the input signal has been set ('input' knob) but before it has undergone any further processing
- **post** - the meters show levels of the output of the plug-in, after the output level has been set ('output' knob)
- if neither **"pre"** nor **"post"** is lit up, the meters show the drive levels of the saturation algorithm

Settings

The sound achieved using the **PSP MixSaturator** is dependent on the type of phonic material being processed. There is, therefore, no universal recipe for obtaining the desired effects. The presets are merely initial proposals for further experimentation.

The main task of the **PSP MixSaturator** is to improve the quality of digital recordings through increasing the middle level and adding harmonics in a similar way to that which happens in high quality analog devices, such as valve processors and analog tape recorders.

Thanks to the additional algorithms, designed with the processing of bass and treble frequencies in mind, a wide range of possibilities is available for adding 'colour' to recordings.

When using the **PSP MixSaturator** in mastering, it is advisable to use the settings for delicately correcting the sound of the recording (the 'mastering' group of presets) and very careful monitoring of the output signal on the best available monitoring equipment.

If this plug-in is used as an output effect while mixing, the 'mix' or 'mastering' presets should be used so that, if required, the sound of the mix can be slightly saturated or corrected. Like in mastering, the employment of the effect requires moderation and careful monitoring, although a lot depends on the kind of material processed.

Working with single tracks provides the greatest opportunities for using the **PSP MixSaturator** for correction. For example, when working with the typical sound of vocals and acoustic instruments, slight 'colouring' introduced through using the 'preamp' programs might be sufficient; however, the sounds of electric and electronic instruments, as well as drums, might require greater saturation (the 'hot valve' and 'hot tape' group) or overdrive and correction (the 'big bass' or 'loop' group).

Using 'Mix' slider allows you to set optimum processing depth without a need to change controls' settings of individual algorithms. It also makes possible mixing clean input signal with deeply processed and distorted signal in any proportion to obtain special effects.

The **PSP MixSaturator** has been equipped with a library of 31 presets. These enable the user to familiarise themselves with the plug-in's possibilities in relation to the processing of different signals, and also serve as a quick guide to particular algorithms and their parameters. To make things easier, the presets have been divided into nine groups:

Group	Application
<i>preamp</i>	individual tracks, vocals, whole mix
<i>mix</i>	for getting punchy and wide mix
<i>mastering</i>	mastering process
<i>contour</i>	the exposition of bass and treble frequencies
<i>big bass</i>	bass instruments and loops
<i>bottom end</i>	drums, whole mix
<i>loop</i>	loops, drums and percussion instruments
<i>hot valve</i>	individual tracks and loops
<i>hot valve</i>	individual tracks and loops

Structure and operation

The analog saturation simulation algorithm

The analog saturation simulation algorithm is used for the treatment of a processed signal as a whole. It enables the user to use one of the seven nonlinearity curves.

There are six curves available, representing the nonlinearity of analog valve processing (Valve1, Valve2, Valve3) and tape saturation (Tape1, Tape2, Tape3), as well as a digital clipping curve.

Each nonlinearity curve gives a different sound for the material processed, as they differ in the arrangement and content of harmonics, as well as the threshold above which distortion becomes audible. The 'drive' slider enables the user to place the input signal on the nonlinearity curve with precision.

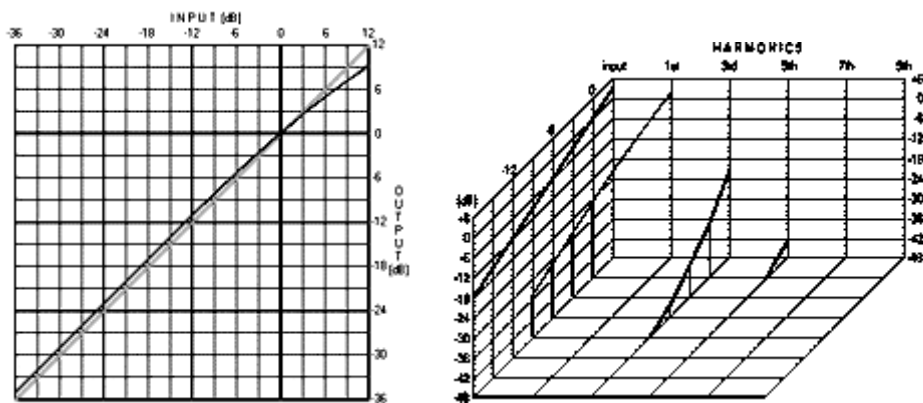
The correct adjustment of the algorithm parameters (settings of the shape and drive sliders) to the processed material makes it possible to obtain a full, clear sound and to modify the signal peaks which are short or too high, similar to the way in which high-quality analog equipment works. It is also possible to obtain purposefully audible distortion thanks to a wide range of settings.

Inappropriate use of the analog saturation simulation algorithm may lead to an undesired and audible reduction of transients and the loss of definition, due to a considerable increase in the level of distortion.

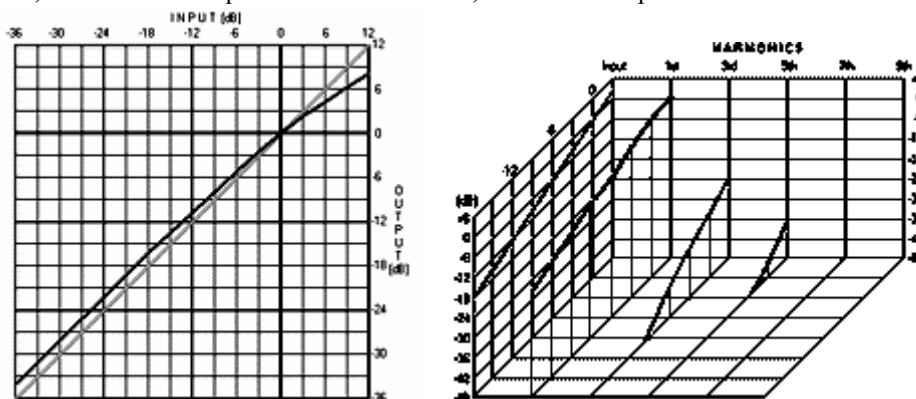
Below are descriptions of particular nonlinearity curves with diagrams of their characteristics. The diagram on the left shows the level of the output signal (output) in relation to the level of the input signal (input); the one on the right shows the increase in the basic compound content and the harmonics in relation to the sinusoidal level of the input signal.

The first three characteristics of the nonlinearity curve represent the way in which valve devices operate at different levels of saturation.

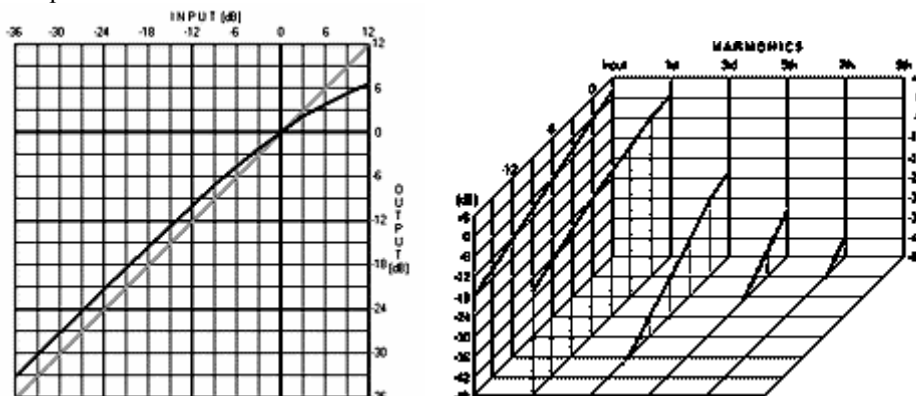
"Valve 1" is a characteristic which introduces very delicate distortion and enables the user to raise the recording level minimally. Its operation leads to an increased clarity in the processed sound and the depth of the sound recording. Thanks to these properties, it is possible to give single tracks, such as vocals or single instruments, or the whole recording, the sound quality of a high-class valve preamp.



"Valve 2" enables the user to achieve a greater amplification than "Valve 1" with the introduction of audible distortion. For this reason it is primarily intended for use in the processing of tracks with marked transients and high peak levels, characteristic of percussion instruments, drums and loops.

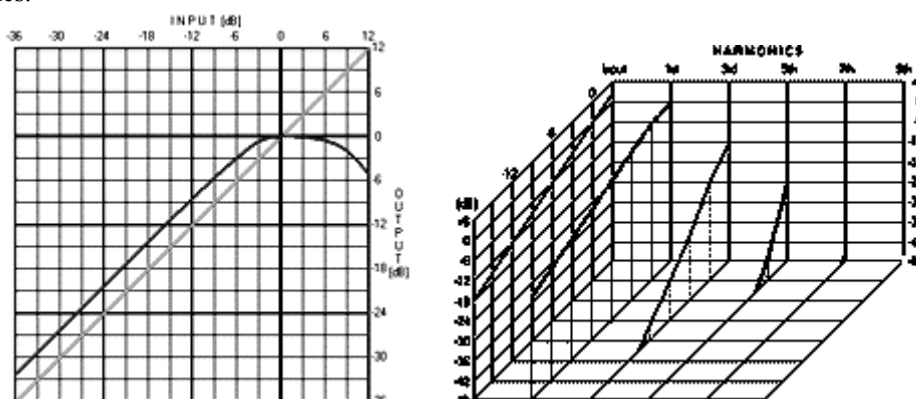


"Valve 3" has been designed to produce the sound of overdriven valve circuits. The use of this kind of sound is dependent on the processed material.



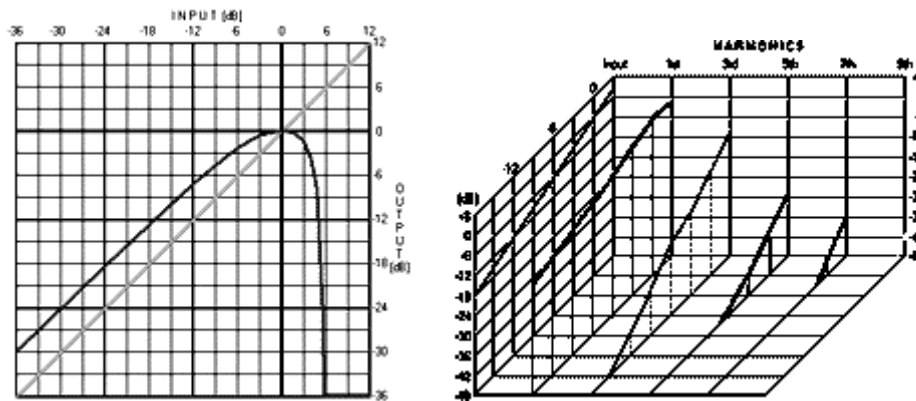
The next three characteristics of the nonlinearity curve are similar to the phenomena produced during the registration and playback of sound signals when using magnetic analog tape. They can be successfully used both for the simulation of slightly saturated sound and the simulation of strongly overdriven tape. Because of the shape of the curves, all three characteristics can raise the recording level by a similar degree. If the overdrive of the algorithm exceeds the maximum level (0db on the indicators and +14db on the meter) the level of the signal will be held or reduced according to further increases in the input signal.

"Tape 1" introduces gentle, barely audible distortion to high input level values. A distinct increase in the distortion content is noticeable in the -6 to 0dB range. The effect of the distortion is concentrated mainly on the third harmonic, which is why "Tape 1" is very good for processing signals with lesser harmonic values, as well as as whole mixes.

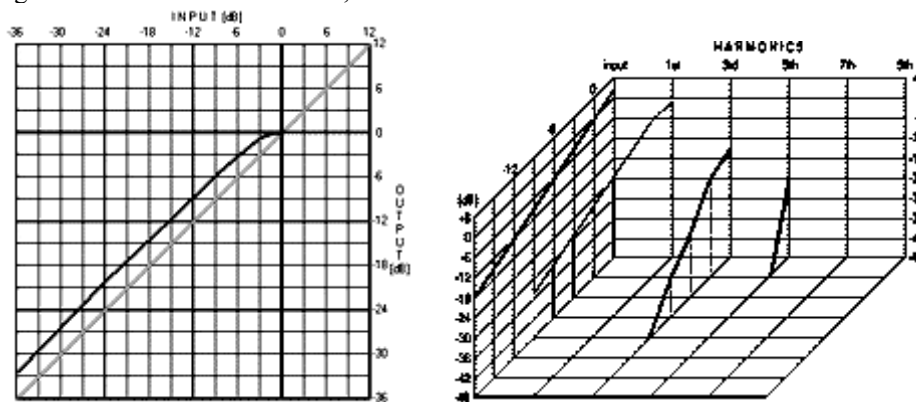


The action of "Tape 2" is more aggressive. Audible distortion is introduced over a greater dynamic range. The effect of the distortion is gently spread over all of the harmonics. A high input-signal level of 3h above 0dB is

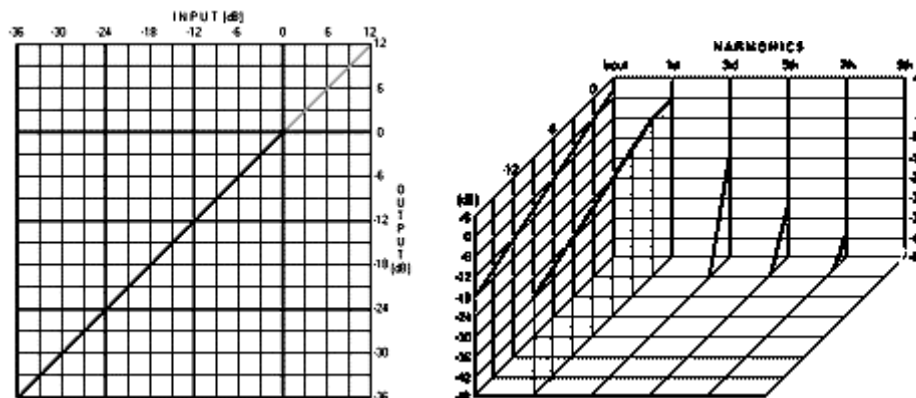
characteristic of "Tape 2" and this is why it is perfect for processing percussion and drum sound, by significantly brightening the transients.



"Tape 3" is characterised by the introduction of distortion only as third and fifth harmonics up to the input level of 0dB. The signal is cut off above this value, due to the increase of all odd harmonics.

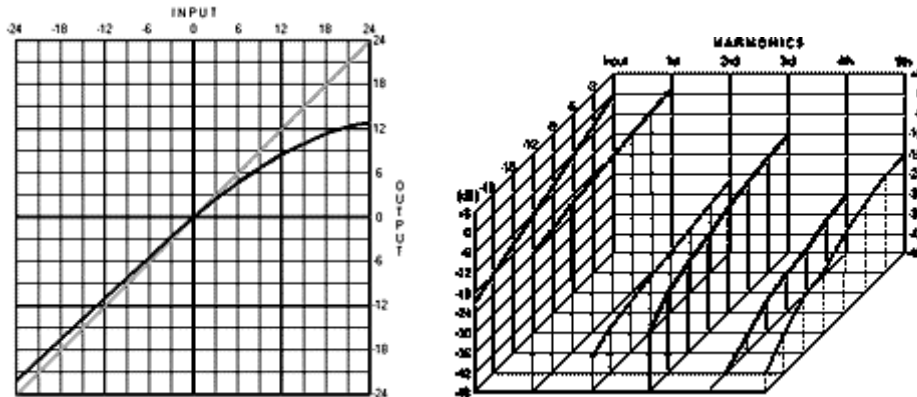


"Digital" is a typical example of digital clipping, which appears, for example, during the overdrive of analog-digital converters (ADC) or a signal recorded on digital media. No distortion is introduced up to the 0dB level, but above this level the distortion increases rapidly and spreads across all of the odd harmonics. The "digital" curve, in distinction from those previously mentioned, does not retain the possibility of increasing the signal level.



The bass frequencies processing algorithm

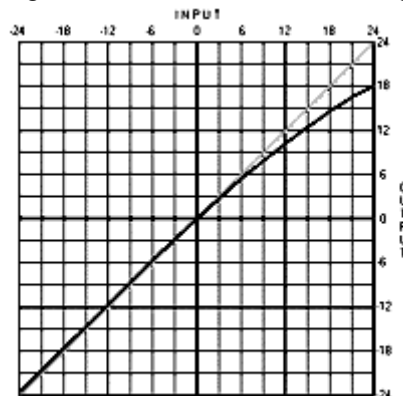
The aim of this algorithm is to add warmth to the sound within the bass frequencies range through adding harmonics and increasing the bass in a recording. The way in which this algorithm affects the sound resembles the action of analog tape, which introduces additional distortion to both even and odd bass frequencies. A wide range of settings makes it possible to obtain a considerably greater range of 'colour'. Technically, this algorithm is a combination of processing with the use of a nonlinearity curve similar to "Valve 2" (which, however, works at higher drive levels) with a special circuit which turns the signal's energy into even harmonic distortion. Below are diagrams which describe the operation of this circuit. The measurement of the harmonics content has been made at the signal level of 0dBFS and the setting of the warmth knob at 100%.



The treble frequencies processing algorithm

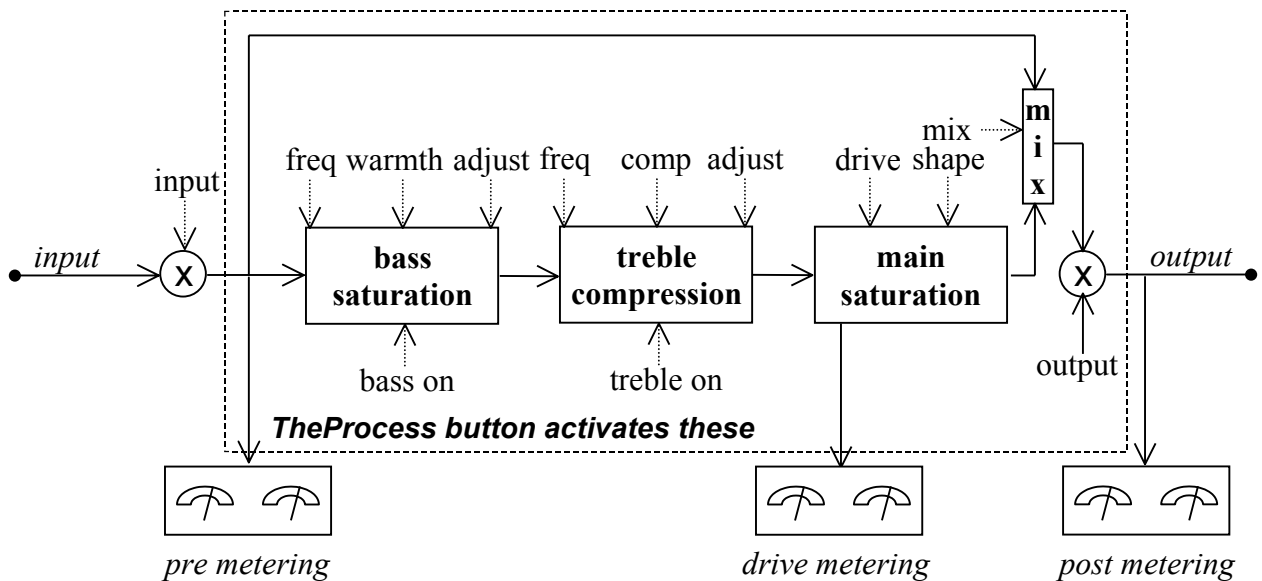
The processing of treble frequencies in the PSP MixSaturator is based on soft-knee compression. The shape of the compression curve is similar to the shape of nonlinearity curve "Valve 1", however, a very high drive level is possible here, due to a high degree of attenuation, which may even exceed 12dB.

Temporary characteristics are automatically selected, based on the set cut-off frequency of the treble frequencies filter. Due to this approach to the processing of treble frequencies, it is possible to simulate the saturation of the tape within this range, without increasing the level of distortion and aliasing.



Like other plug-ins from the MixPack 1.5 series, the PSPMixSaturator's 'mix' control can be used for mixing the input signal with the processed signal in the desired proportions. This function allows for quick and comfortable setting of the optimum depth in all plug-in algorithms. This regulation is before output level control.

Block diagram



Description of terms used in the block diagram for the **PSP MixSaturator** processor

<i>input</i>	plug-in input: regulates the input signal
<i>output</i>	plug-in output: regulates the output signal
<i>bass saturation</i>	bass frequencies processing algorithm
<i>freq</i>	cut-off frequency of the filter
<i>warmth</i>	addition of warmth to bass frequencies
<i>adjust</i>	content of bass frequencies
<i>bass on</i>	turning on the processing of bass frequencies
<i>treble compression</i>	treble frequencies processing algorithm
<i>freq</i>	cut-off frequency of the filter
<i>comp</i>	treble frequencies compression
<i>adjust</i>	treble frequencies content
<i>treble on</i>	turning on the processing of treble frequencies
<i>main saturation</i>	analog saturation simulation algorithm
<i>drive</i>	drive level of the saturation simulation algorithm
<i>shape</i>	selection of the nonlinearity characteristics
<i>mix</i>	regulation of the proportion of input signal to processed one
<i>pre metering</i>	measuring signal levels after regulation of the input level
<i>drive metering</i>	measuring drive levels of the saturation simulation algorithm
<i>post metering</i>	measuring of the output signal levels
<i>Process button activates these sections</i>	processor elements within the frame are active if the "process" button is on

PSP MixPressor

General information

PSP MixPressor is an advanced compressor designed for the final processing of whole mixes or single tracks. Combining the advantages of digital signal processing with algorithms which emulate the operation of analog circuits, it enables the user to obtain a sound quality typical of compressors with valve and optoelectronic circuits without any unwanted noise or distortion.

The **PSP MixPressor** offers a wide range of features, providing a unique universality:

- » This compressor works with a 'soft knee' which lessens the negative effects connected with the operation of the system.
- » The level detector can operate in two modes: peak mode and RMS mode, which enables the user to adjust the time characteristics to the material processed.
- » The times of the attack and release can be regulated across a wide range, or an automatic setting can be chosen. The regulation of the 'hold' makes it possible to regulate the character of the bass-frequency sound.
- » The side chain signal has a smoothly-regulated 'bell-type' filter, thanks to which 'pumping' can be limited, or the compressor can be used as a 'de-esser'.
- » The PSP MixPressor also contains a limiter which can operate as a warm-sound peak reducer, useful during mastering, or as a limiter-saturator with a sharper sound, which is excellent for brightening-up the sound of percussion loops.
- » The plug-in is equipped with unique meters which simultaneously indicate the peak and VU levels before or after compression, as well as the mean gain reduction.

Features

- » compressor with 'soft-knee' characteristics,
- » peak or RMS level detection,
- » automatic and manual timings for attack and release,
- » filter in the side-chain circuit,
- » limiter-saturator,
- » unique peak, VU and gain-reduction meters,
- » mixing of input signal with processed one in the desired proportions,
- » contains a set of factory presets.

Applications

- » 'soft' compression of whole mixes during mastering,
- » compression of vocals and solo instruments,
- » compression of percussion instruments,
- » compression of an unrestricted types of tracks or groups during mixing,
- » soft knee limiting.

Specifications

Plug-in type: Plug-in VST, PC and Mac OS version

Internal signal processing: 32-bit, floating-point

Internal delay: depends on settings - in the range of 0 to 25ms

Processing speed: below 12% of the computation power of the PII 333MHz or G3 300 MHz processor with all algorithms set on

Controls

The operation of the parameter settings of the PSP MixPressor plug-in has been standardized to be compatible with other elements of the VST system.

- » Control settings are changed by clicking or clicking and dragging.
- » Clicking and dragging the slider while holding down the 'shift' key ensures precision in changing the slider settings.
- » Knobs operate in a linear mode by default, however you could press 'alt' key to operate in circular mode; in Cubase 5.0 and Nuendo 1.5 host applications the knob operation is set by the superior host application.
- » Clicking on a knob or a slider while holding down the 'Ctrl' key will restore the standard setting.
- » Clicking on the name of the plug-in causes the window 'information about' to appear.



Buttons

[process]

activates processing in the plug-in. If this button is not lit up, the only active controller remains the 'input' knob, which regulates the gain of the input signal

[del] (delay)

activates the internal delay which reduces the signal peaks during the attack

[scl] (side chain listening)

activates the listening-in the side-chain signal

[rms] (root mean square)

if this button is lit up, it activates RMS detection of the signal level characteristic for optoelectronic circuits. If the button is not lit up, it operates as a peak detector

[sat/lim]

switches the limiter between a warm-limiter function and a limiter-saturator function, or shuts the limiter off altogether. The limiter operates at an output level of 0dBFS.

[oo/o]

stereo/mono switch. It is set automatically when the plug-in is turned on; it can also be set manually.

Knobs

[input]

regulation of the input signal gain

[output]

regulation of the gain of the output signal before the limiter section

[freq]

frequency of the filter in the circuit which drives the compressor

[Q]

quality factor of the filter - the higher the Q factor, the steeper the slopes of the filter

[attack]

attack time; modes: fast automatic, slow automatic and manual regulation

[hold]

time for which the level is held

[release]

release time; modes: fast automatic, slow automatic and manual regulation

[make-up]

restoring the level after compression; modes: automatic restoration of the levels and manual regulation

Sliders**[compress]**

regulation of the depth of compression

[slope]

regulation of the bend of the compression curve

[mix]

regulation of the input to processed signal ratio. It allows to set processing depth of plug-ins' algorithms precisely

Peak level, VU and gain reduction meters with an overdrive signaling system

The **PSP MixPressor** meters are complex measuring devices, which are used to establish the work-levels of the algorithms in the plug-in, the levels of the output signal and the dynamic accuracy of the processed signal.

The meters have been given a logarithmic scale. The difference between the peak value and VU for typical music signals is 14dB, hence the maximum value on the scale is +14 which is equal to 0dbBFS, with the value 0 being equal to the VU level of a correctly made recording.

red needle indicator

this indicator shows the peak level of signals, which is measured with an accuracy to one sample. The maximum level is held for 300ms.

black needle indicator

this indicator shows the middle-level VU of the signal. The integration time is 300ms. The indicator shows a level equal to the peak signal for the established sinusoidal signal.

short needle blue indicator

this indicator shows the mean gain reduction of the signal during compression; this indicator does not indicate the gain reduction which results from the work of the limiter circuit.

overdrive' signal

LED diodes light up on reaching or exceeding +14 (0dbBFS) on the scale or when the work of the limiter is too deep - i.e. audible.

[pre/post] switch

meters mode switch

- **pre** - the meters show the level after the input signal gain level has been set (the 'input' knob), but before further processing of the signal
- **post** - the meters show the signal as it exits the processor after the output signal level knob and limiter section.

Settings

The **PSP MixPressor** is a compressor designed for processing single tracks, groups of tracks or whole mixes when high, unimpaired quality and warm sound are required. In addition, the **PSP MixPressor** enables the user to employ a wide range of settings, while ensuring precise control of the effects.

The **PSP MixPressor** contains a library of factory presets, which can be a good starting point in the search for suitable sound effects. After selecting a preset suitable for the material to be processed, the input signal level should be adjusted by means of the 'input' knob or the setting of the 'compress' slider should be corrected. It should be remembered that a deep compression combined with a prolonged attack time produces an increase in the level on entering the limiter. If you are using the automatic make-up level setting, the input level of the limiter should be reduced using the 'output' knob.

The 'Mix' slider enables the user to obtain very interesting compression effects. Setting this parameter below 100% decreases the influence of compressor operation on the signal peaks by moving the point of its operation to the middle dynamic range. Using this slider enables the application of the **PSP MixPressor** in whole mixes without destroying transients, with simultaneous dynamic modulation and a general increase in the average level. It also enables the user to obtain what is known as the 'American drum sound', which in traditional recording studios is achieved by mixing a deeply-compressed signal with its non-compressed version.

The search for special compression effects requires a deeper knowledge of the operation of the **PSP MixPressor** and experience in working with compressors.

The side-chain filter has a particular significance for the compressor sound. Its influence on signal processing increases with the increase in the value set using the 'Q' knob. The 'SCL' button should be turned on in order to tune the side-chain filter. When using settings designated for the operation of the compressor only at certain frequencies (e.g. for a 'de-esser'), it is advisable not to use the automatic make-up gain setting, which, in this case, might cause the limiter/saturator to work too deeply, or overdrive the plug-in output.

The filter also enables the user to operate the compressor as a 'de-esser' (when the 'Q' value is high and the 'freq' value is between 5 and 10 kHz), or to lessen the 'pumping' effect, which can be caused by a high bass-drum level (a 'Q' value between 0 and 0.3, and 'freq' around 500Hz).

The search for interesting attack and release-time settings can produce fascinating results. For example, you can artificially produce a sharper attack of the 'pad' or 'string' track by using a prolonged attack-time setting (300 - 1000ms), short release time (10ms), and compression depth of above 50%, not forgetting to turn off the automatic make-up level by setting the knob in the 0dB position; the 'del' button should be turned off as well.

Alternatively, the 'pumping' effect can be achieved by changing automatic attack and release times in favor of '12 o'clock' settings (attack around 30ms and release around 350ms); the compression should be set at above 50%, 'DEL' and 'RMS' turned off, and 'Q' set below 0.2.

The **PSP MixPressor**'s limiter section can also be used in many ways. One of its traditional uses (the 'LIM' setting) is in the elimination of excessively high signal peaks, which might overdrive plug-ins or other elements of the side chain after the plug-in. LED diodes signal the overdrive only when the operation of the limiter is too deep, which is why they are useful if audible distortion is to be eliminated from the output signal.

When the limiter is set to deep work it would emulate the sound of saturated tape. In this case, the 'LIM' or 'SAT' mode should be chosen, depending on the processed signal and the sound desired. The 'LIM' setting is conducive to generating gentle distortion, which is excellent for bass sounds, pads and strings; the 'SAT' setting accentuates transients by adding brightness to loops or percussion instruments.

In both cases it should be remembered that the compression level in the limiter section is regulated by the 'output' knob, which is situated before the limiter in the side chain. The maximum output level of the limiter does not exceed 0dBFS.

When using the **PSP MixPressor** in pre-mastering, it is advisable to use settings which create a gentle compression within the range of several dBs and with the limiter producing single flashes at most on the LED diodes. 'mix1' and 'mix2' settings, designed for warming the sound and for slight increasing the level as well as for the dynamic arrangement of a ready mix, may be useful here. You can also use the 'so long' preset in order to even up the level for the whole recording or part of it.

'Mix' group settings should be employed if the plug-in is to be used as a final effect while mixing.

Another interesting solution is to use the **PSP MixPressor** in combination with the **PSP MixSaturator**. In this case, the **PSP MixSaturator** should be used before the **PSP MixPressor** in the signal path. Employing both plug-ins simultaneously at their gentle settings enables the user to obtain very interesting effects and to provide the whole mix with a 'produced' sound.

The **PSP MixPressor** has been equipped with a library of 37 presets. These enable the user to familiarise themselves with the plug-in's possibilities in relation to the processing of different signals, and also serve as a quick guide to particular algorithms and their parameters. To make things easier, the presets have been divided into eleven groups.

Group	Application
<i>universal</i>	universal setting for further experimentation
<i>mix</i>	realization of mixes, pre-mastering
<i>track</i>	processing of single tracks
<i>vocal, de-esser</i>	vocal tracks
<i>guitar</i>	guitars
<i>bass</i>	bass instruments
<i>brass</i>	brass instruments
<i>so long</i>	leveler
<i>delay FX</i>	delayed release
<i>drums</i>	percussion instruments, drums, loops
<i>classic*</i>	single tracks and whole mix

***Attention:** the classic group contains a set of presets which simulate the operation of studio analog compressors. Although the names of the presets may be associated with classic brands of compressors, it should be remembered that the **PSP MixPressor** is a universal plug-in, which is why it does not claim to be a replica of any of these devices. These presets are only an attempt to simulate the time and dynamics characteristics typical of classic analog compressors. They have been designed to find the sound sought by setting the compression depth with the 'compress' slider, and the after-compression signal regeneration with the 'make-up' or 'output' knobs.

It should be remembered that the 'make-up' knob changes the level before, and the 'output' knob after, the 'Mix' slider, which will produce different sound results if the 'Mix' slider is at a setting other than 100%.

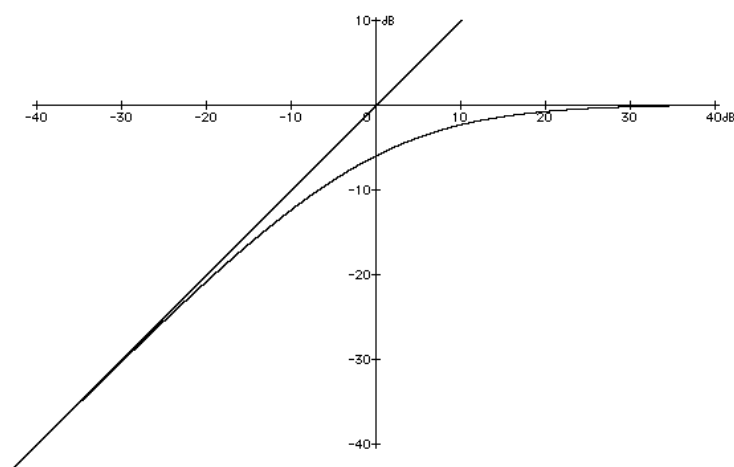
Structure and operation

The **PSP MixPressor** consists of two modules: the compressor and the limiter. Details of the internal structure are presented in the block diagram of the plug-in.

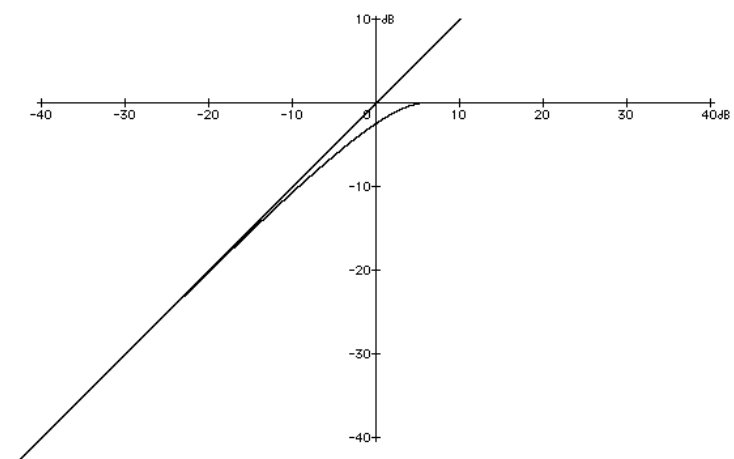
Compressor

The work of the compression algorithm in the **PSP MixPressor** plug-in is based on a gently bent compression curve, the range of which can be easily and smoothly adjusted.

The first diagram shows the gentle compression characteristic which is obtained when the 'slope' slider is in the down position. This type of characteristic is useful for the gentle sound 'warming' of solo instruments or vocals. The compressor when set in such a way will operate within a wide dynamic range and, therefore, the articulation will be delicately boosted.



The second diagram shows the characteristic when the 'slope' slider is in the up position. The transition from the uncompressed level to the limited level takes place quickly but smoothly, which is why this characteristic may be called the characteristic of the 'soft-knee' limiter. This type of setting is especially useful for processing percussion instruments, or when using the compressor as a gentle limiter.



In most cases, a low position of the 'slope' slider should produce a satisfactory warm sound in the processed material.

The characteristics presented have been determined without taking the operation of the 'compress' slider into consideration, which is responsible for setting the threshold of the compression in the range from +12 to -48 dBFS.

In order to make working with the compressor easier, an automatic 'make-up' option has been employed. If the signal after compression with the 'make-up' knob set at 'auto' needs a slight level adjustment, this can be achieved by means of the 'output' knob, which precedes the limiter in the signal path.

When necessary (for example, while using the compressor as a 'de-esser') the make-up level after compression with the 'make-up' knob within the range 0 to +48dB, can also be set manually.

Apart from the static compression characteristic, a proper choice of attack, hold, and release times is also important for the sound of the compressor.

The level detector is the first element responsible for temporary characteristics of the compressor. The **PSP MixPressor** enables the user to set the mean square (indicated as 'RMS') or peak detection. RMS detection is approximate to perceptual volume reception and peak detection facilitates more precise control of the dynamics of the processed signal, which is often useful while processing tracks which contain transients (cymbals, for example). The final choice of either option should depend on the type of processed signal and the sound required.

The attack time is decisive in maintaining the sound character of tracks containing transients. It includes two automatically regulated settings - fast and slow - which are satisfactory in most situations. However, the change of the attack time can also be smoothly regulated within a very wide range (from 0.1 to 1000ms).

An internal optional delay, turned on using the 'del' button, has been employed in the **PSP MixPressor** plug-in in order to reduce sharp signal peaks during the attack. The delay depends on the time of the attack and the mode of the level detector.

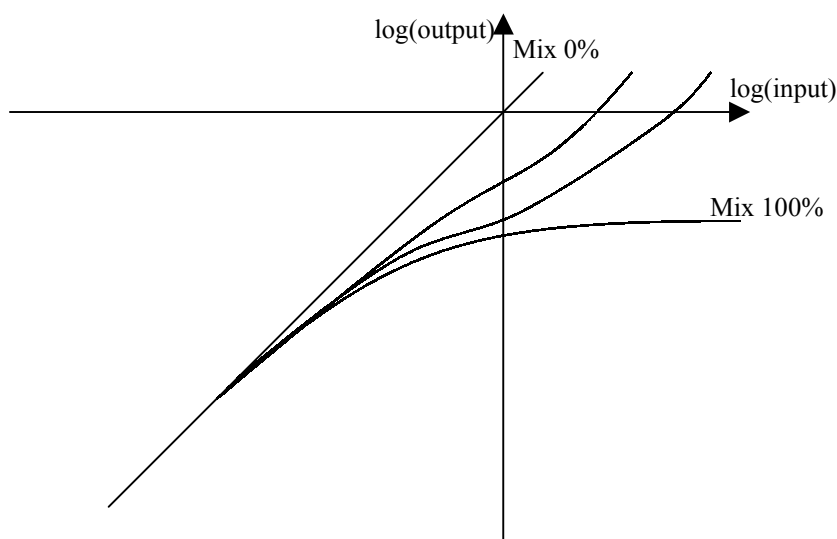
The setting of the release time is responsible for a medium increase in the recording level. If the release time is very short, the compressor will operate more like a limiter, and the sound will resemble a highly saturated or overdriven analog tape; and long release times will cause the compressor to operate as a leveler, which is used for general leveling of the dynamics of particular parts of a mix. When the compressor is in normal operation, medium times are most common (50 to 1000ms), facilitating the proper arrangement of particular tracks in a mix without audible distortion.

The release time, similar to the attack time, can be set as one of two automatic settings (fast or slow) or manually within a range of 10ms to 10s.

If the processed signal requires fast release times, during which clearly audible and unpleasant distortion can appear, some of the inconvenience can be reduced by a proper setting of the hold time, which thus limits the effect of modulation on individual bass waves.

The last element of the compressor is a tunable filter with a 'bell-type' characteristic in the side-chain signal path. It enables the user to reduce the 'pumping' which is caused, for example, by a strong bass drum signal, to use the compressor as a 'de-esser', or to obtain other interesting effects.

Using the 'Mix' slider during compression enables additional control over the dynamics curves. When this parameter is set at 100%, the dynamics curve acts as a limiter curve with a very soft knee regulated by the slope parameter, but setting the 'Mix' parameter below 100% causes an increase in the dynamics curve for high levels. The picture below shows the influence of the 'Mix' parameter on the general operation of the compressor circuit.



Limiter

The **PSP MixPressor** limiter has been designed in such a way that the output signal does not exceed 0dB. The input signal level of the limiter is set by means of the 'output' knob situated before the limiter in the circuit.

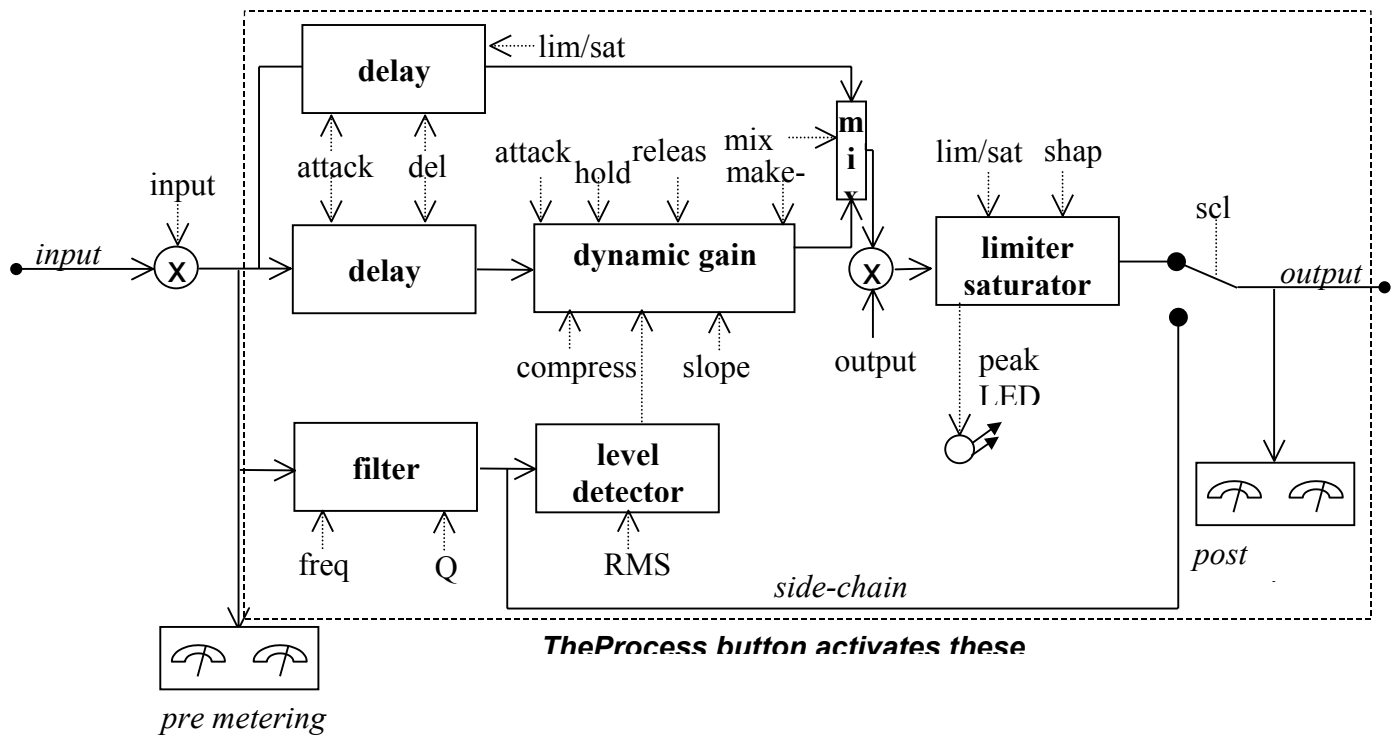
The limiter enables the user to limit sharp signal peaks which appear on exiting the compressor. It also facilitates the maximization of the signal level without exceeding the 0dB level. The audible work of the limiter, which is too deep, is indicated by the LED diodes in the meter section of the plug-in.

The limiter can operate as a limiter with a warm sound ('lim' mode). In this mode an additional permanent delay (2.2ms) is activated, which facilitates permanent penetration of the signal in order to correctly select the compression depth of the limiter. This mode performs well with most instruments and mixes.

The second setting of the limiter changes its characteristics so that the sound more closely resembles a combination of the limiter and saturator. The effect of the limiter operation in this mode is a sharper sound which is perfect for processing drums, loops and other instruments which contain transients and a large number of harmonics. Use of the limiter in the 'sat' mode does not cause any delay.

Like other plug-ins from the MixPack 1.5 series, the PSP MixPressor's 'Mix' control can be used for mixing the input signal with the processed signal in the desired proportions. This function allows for quick and comfortable setting of the optimum compression depth and shape. This regulation is after 'make-up' and before 'output' control of the plug-in and does not influence directly to the operation of limiter/saturator algorithm.

Block diagram



Description of terms used in the block diagram for the **PSP MixPressor** processor.

<i>input</i>	plug-in input; the regulation of the input signal gain
<i>output</i>	plug-in output; the regulation of the output signal gain
<i>DEL</i>	activation of the internal delay
<i>Q</i>	quality factor of the filter
<i>freq</i>	resonant frequency of the filter
<i>RMS</i>	switching between RMS and peak level detection
<i>attack</i>	attack time
<i>hold</i>	time for which the level is held
<i>release</i>	release time
<i>compress</i>	depth of compression
<i>slope</i>	sharpness of the compression knee
<i>make up</i>	adjustment of the level after compression
<i>lim/sat</i>	activation of the limiter/saturator
<i>SCL</i>	listening into the side chain
<i>mix</i>	regulation of the proportion of input signal to processed one
<i>Side Chain</i>	the side chain which drives the level detector
<i>pre metering</i>	measurement of signal levels after regulation of the input level gain
<i>post mastering</i>	measurement of output signal levels
<i>Process button activates these sections</i>	processor elements within the frame are active if the 'process' button is on

PSP MixTreble

General information

PSP MixTreble has been designed for processing a range of treble frequencies. Its universality and great possibilities in creating sound are ensured by these expanded sections:

- » the hiss remover section which allows a decrease in the hiss content or in the undesired reverberation of a room in the range of treble frequencies,
- » the transient section designed for stimulating flattened transients,
- » the enhancer section which makes it possible to increase the spatiality of the processed material,
- » the harmonic section designed for widening the frequency range by enriching the signal in lacking harmonics.

Specifications

- » four algorithms designed to provide processing adequate for different aspects of sound in the high frequency range,
- » wide range of possible sounds,
- » switchable soft clipping algorithm which prevents output level from exceeding 0dBFS,
- » principles of algorithms based on analog devices,
- » mixing of input signal with processed one in the desired proportions,
- » contains a set of factory presets.

Applications

- » processing single tracks for better presence in the mix,
- » processing any kind of tracks or groups for removing the noise floor and better stereo image
- » processing mixes or archival material for overall treble improvement - the better S/N ratio, detailed sound, clear articulation and breath.

Specifications

Plug-in type: Plug-in VST, PC and Mac OS version

Internal signal processing: 32-bit, floating-point

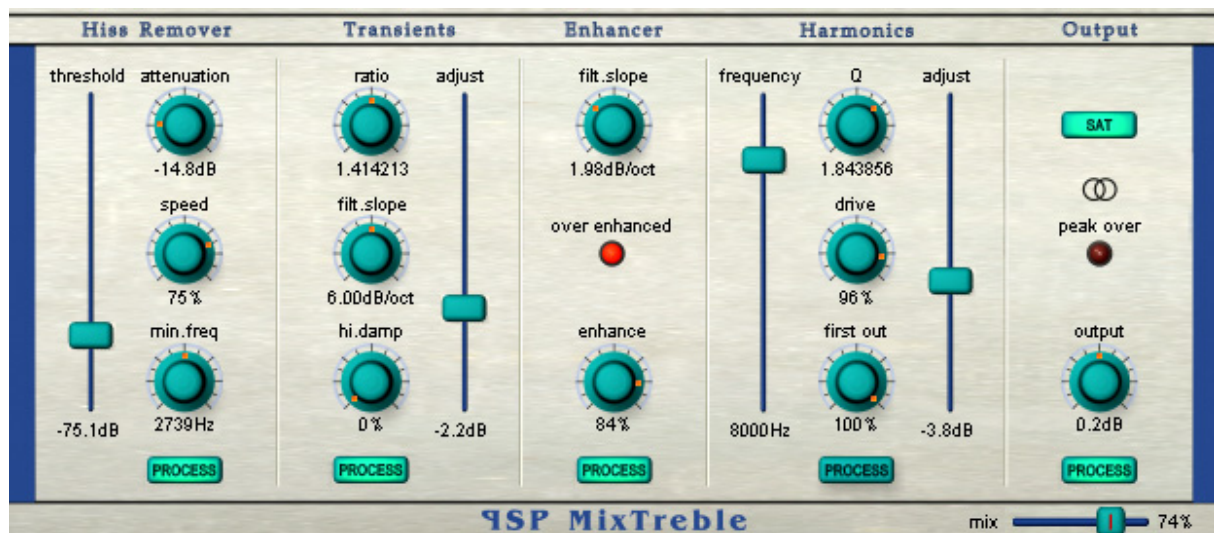
Internal delay: 256 samples with 'hiss remover' turned on, 0 samples in other case

Processing speed: below 30% of the computation power of the PII 333MHz or G3 300 MHz processor with all algorithms set on

Controls

The operation of the parameter settings of the PSP MixPressor plug-in has been standardized to be compatible with other elements of the VST system.

- » Control settings are changed by clicking or clicking and dragging.
- » Clicking and dragging the slider while holding down the 'shift' key ensures precision in changing the slider settings.
- » Knobs operate in a linear mode by default, however you could press 'alt' key to operate in circular mode; in Cubase 5.0 and Nuendo 1.5 host applications the knob operation is set by the superior host application.
- » Clicking on a knob or a slider while holding down the 'Ctrl' key will restore the standard setting.
- » Clicking on the name of the plug-in causes the window 'information about' to appear.



Hiss Remover

[threshold]

The signal threshold in the control side-chain of the filter, below which the unwanted signal is defined.

[attenuation]

The depth of attenuation of the filtered high-frequency range. In order to ensure smooth noise reduction, it is recommended that the parameter is set within a range from -6 to -12dB.

[speed]

the speed of the filter cut-off frequency, which depends on the processed material. In most cases, the mid. position of this knob gives good results. When processing material with a high transient level, it is advised to set the parameter below 50%. For sounds with soft attack and decay, "slow strings" for example, the value of the parameter should be increased.

[min.freq.]

Indicates a minimal frequency for the dynamic filter. In most cases, the mid. position guarantees satisfactory noise reduction, without deep high-frequency modulation.

[Process]

Activates this section.

Transients

[ratio]

Smoothly regulated expansion ratio between 1:1 and 2:1.

[filt.slope]

Sets the high-pass filter slope within the range from 0 to 12dB/oct.

[hi.damp]

Damping of extremely high frequencies during high-frequency filtering to avoid "overbrightness" of this range.

[adjust]

Adjusts the processed signal content in the output signal.

[Process]

Activates this section.

Enhancer**[filt.slope]**

Sets the high-pass filter slope within the range from 0 to 6dB/oct.

[over enhanced]

Indicates the possibility of an overenhanced effect due to the high level of the side (S) component in the output signal.

[enhance]

Adjusts the depth of the stereo enhancement effect. The 0% setting does not cause any changes in the original signal.

[Process]

Activates this section.

Harmonics**[frequency]**

Sets the filter center frequency.

[Q]

Sets the filter quality factor.

[drive]

Sets the depth of the drive effect for the filtered signal and is responsible for the amount of generated harmonics.

[first out]

Enables the possibility of removing the fundamental component of the signal and leaving only the harmonics for further regulation.

[adjust]

Sets the amount of generated harmonics in the output signal.

[Process]

Activates this section.

Output**[Sat]**

Activates the soft-clipping algorithm for the output signal.

[oo/o]

Switches the plug-in between mono and stereo processing.

[peak over]

Indicates a peak-signal level higher than 0dB if the soft-clipping algorithm is off, or the almost full saturation of the algorithm, if it is on.

[output]

Sets the output signal level. It comes before the soft-clipping algorithm in the signal path.

[Process]

Activates all sections of the plug-in.

Settings

The **PSP MixTreble's** high quality of signal processing means that it can be used in a wide range of applications. It can be used for improving sound definition, boosting the characteristic features of acoustic instruments and drums, revitalizing archival records, and also to enhance the overall sound quality of ready mixes.

The **PSP MixTreble** allows extending dynamic range, sharpness, clarity and spaciousness of processed material however results depends on both character of a signal and the way plug-ins' algorithm are used.

Hiss Remover

This section is designed to reduce the noise content in the treble frequencies. However it also allows to reduce the perceived level of background sounds as well as reverberation of a room. This is also possible to use this section to deeply process the sound of drum loops or synthesizer tracks. The way this algorithm works resembles the operation of similar analog devices. Its main operating element is a dynamically tuned low-pass filter, designed for filtering out the unwanted high-frequency signal.

In order to set the algorithm properly it is recommended that you:

- » keep only this section on ,
- » set "attenuation" at -∞ dB, "min.freq" at 500Hz, "speed" in the mid. position and "threshold" at -96dB,
- » gradually increase the threshold value until you achieve an audible reduction in hiss, without a significant loss of transients and the desired high tones,
- » set "min.freq" to reduce the high-frequency modulation effect,
- » set "attenuation" within a range from -12 to -6dB,
- » adjust 'attenuation' parameter between -12...-6dB.

Transients section

This section is designed for enhancing transients which have been flattened by poor-quality analog equipment. The operation of this section is based on the high-frequency compander. Due to this algorithm, it is possible to revitalize tracks whose details have been softened, as well as to improve definition in the medium and high-frequency range, without an increase in the noise content.

In order to set the algorithm properly, it is recommended that you:

- » keep this section and the "Hiss Remover" section on,
- » set "ratio" and "filt.slope" in the mid. position, "hi.damp" at 0%, and "adjust" at -∞ dB,
- » gradually increase the "adjust" parameter value to achieve a satisfactory increase in the definition and dynamics,
- » adjust the "hi.slope" parameter value: if it is necessary to revitalize higher frequencies, the slope of the filter should be increased,
- » correct the expansion depth using the "ratio" parameter,
- » in case of overbrightness, it should be reduced by increasing the "hi.damp" parameter value.

Enhancer section

This section is designed for enhancing the stereo base or increasing the spatiality of high frequencies. It works using the high-pass filter and a controlled XY->MS->XY matrix. It can be active only during stereo-signal processing.

In order to set the algorithm properly, it is recommended that you:

- » set the "filt.slope" knob in the middle position,
- » set the "enhance" knob at 0%,
- » find a setting for "enhance" which does not cause too large an increase in the value of the spatial component of the signal,
- » correct the "filt.slope" parameter setting so as to find a suitable frequency range for the operation of the effect.

Harmonics section

This section is designed for generating additional harmonics for improving clarity and definition, or adding presence to a track. The structure of the algorithm also allows band distortion of a signal. This algorithm has been equipped with a wide-range tunable bell-type filter. The filtered signal is sent to the harmonics generator and then to the fundamental component-removing filter. The harmonics generator generates odd and even harmonics.

In order to set the algorithm for typical uses, it is recommended that you;

- » set the "Q" and "drive" parameters in the mid. position,
- » set "first out" at 0%, and "adjust" at 0dB,
- » find the medium frequency of the filter which can be a base for generating desired harmonics,
- » filter out the fundamental component of the signal by setting the "first out" knob at 100%,
- » set the desired harmonic level with the "adjust" slider.

The PSPMixTreble has been equipped with a library of 38 presets. These enable the user to familiarise themselves with the plug-in's possibilities in relation to the processing of different signals, and also serve as a quick guide to particular algorithms and their parameters. To make things easier, the presets have been divided into eleven groups:

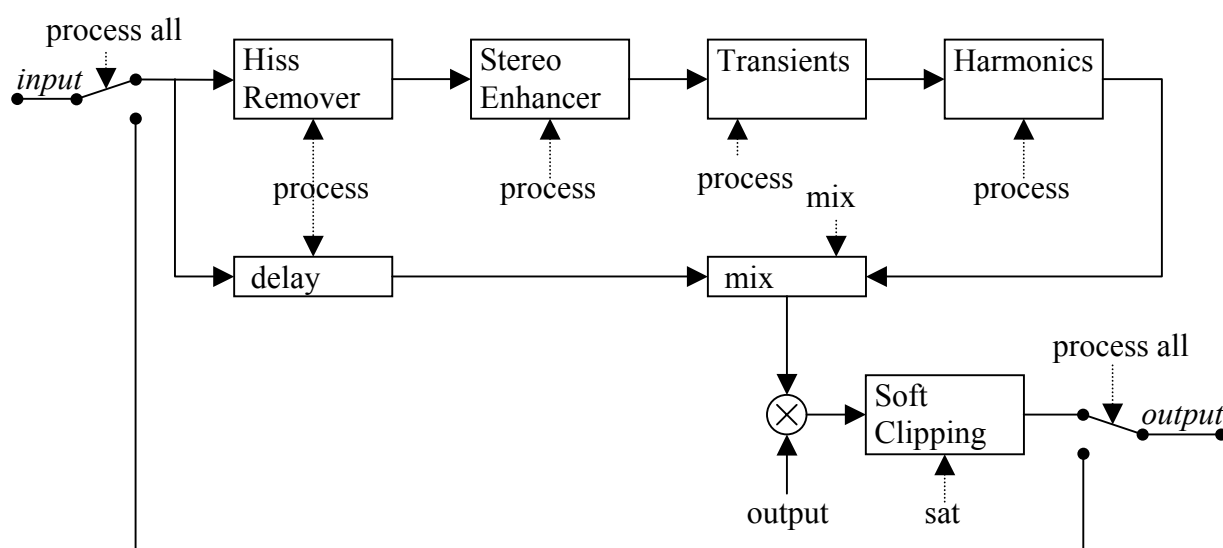
Grupa	Zastosowanie
<i>mix</i>	mixing
<i>acc guitar</i>	acoustic and classic guitars
<i>vocal</i>	vocals and choirs
<i>drums</i>	drums and percussion
<i>piano</i>	pianos and electric pianos
<i>violino</i>	violin and strings
<i>universal</i>	various material
<i>hiss rem</i>	noise and room reverberation canceling
<i>transients</i>	transient exposition, dynamic high frequency equalization
<i>stereo enh</i>	spatial enhancement of stereo tracks
<i>harmonics</i>	adding new harmonics to a track

Structure and operation

The PSPMixTreble contains a set of algorithms designed for processing treble frequencies:

- » the hiss remover section which allows a decrease in the hiss content or in the undesired reverberation of a room in the range of treble frequencies,
- » the transient section designed for stimulating flattened transients,
- » the enhancer section which makes it possible to increase the spatiality of the processed material,
- » the harmonic section designed for widening the frequency range by enriching the signal in lacking harmonics,
- » the soft-clipping algorithm, which enables protection of the plug-in output against exceeding 0dBFS.

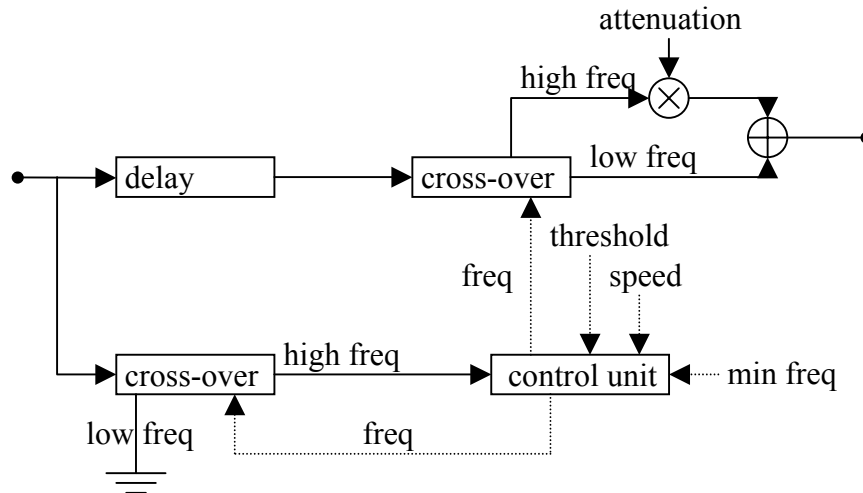
A general block-diagram of the MixTreble



<i>input</i>	wejście pluginu
<i>output</i>	wyjście pluginu, regulacja wzmocnienia sygnału wyjściowego
<i>Hiss remover</i>	sekcja redukcji szumu
<i>Stereo Enhancer</i>	sekcja poszerzania stereofonii
<i>Transients</i>	sekcja uwydatniania transientów
<i>Harmonics</i>	Sekcja generowania harmoniczných
<i>delay</i>	opóźnienie
<i>Soft clipping</i>	łagodne obcinania szczytów sygnału
<i>mix</i>	regulacja proporcji pomiędzy sygnałem wejściowym a przetworzonym
<i>process all</i>	włącza cztery główne algorytmy plug-inu
<i>process</i>	włącza poszczególne sekcje plug-inu

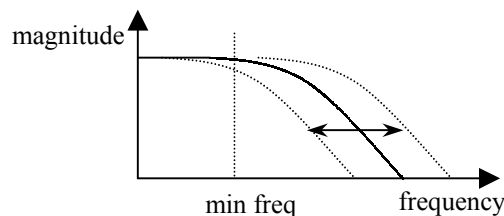
Hiss Remover section

This section is designed to reduce the noise content in the treble frequencies. The way this algorithm works resembles the operation of similar analog devices. Its main operating element is a dynamically tuned low-pass filter, designed for filtering out the unwanted high-frequency signal (hiss noise, background noise or room reverberations). The way this algorithm operates is presented below.



The structure of this dynamic algorithm with the precise control of its parameters ensures optimum processing for most sound material. The input signal undergoes a constant delay of 256 samples in order to safeguard against the transients attenuation.

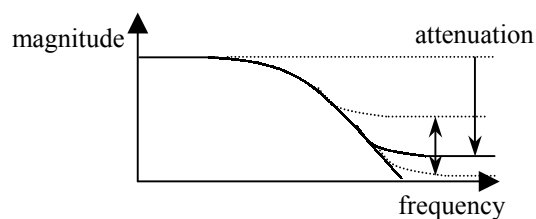
The cut-off frequency of the filter changes depending on the content of treble frequencies above the current cut-off point (see graph below). If the content of treble frequencies (being filtered) is higher than the threshold, then the cut-off frequency of the low-pass filter is moved proportionally towards higher frequencies. If the level of the filtered frequency band is lower than the threshold, the filter tunes to lower frequencies.



Both the scaling of the dynamic of change in tuning in the filter, as well as the minimal filter frequency, are subject to additional control in order to ensure the universality of the algorithm. Regulation of filter speed ensures its adjustment to the dynamic of the signal, and regulation of minimal frequency protects the system from deadening the sound too much, which could impair the quality of the signal in the case of transients.

The hiss-remover algorithm also enables the setting of the depth of 'attenuation'. This parameter enables precise control of the operation depth of the algorithm. With the help of this parameter, you can use the effect both for individual tracks with a high level of hiss and whole mixes which require only slight correction.

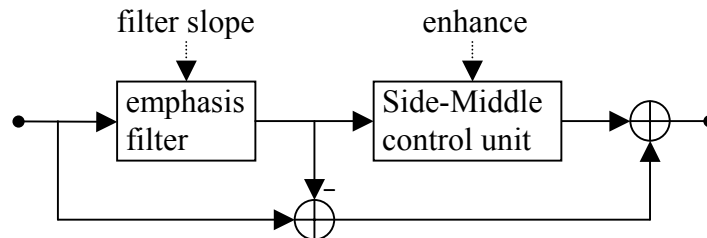
A diagram of the effects of the parameter is shown in the picture.



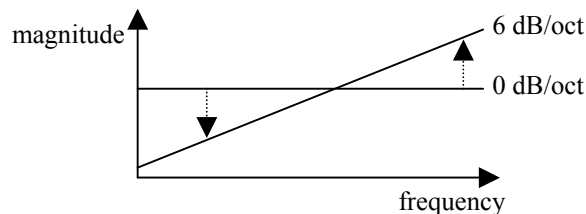
Enhancer section

This section is designed for enhancing the stereo base or increasing the spatiality of high frequencies. It works using the high-pass filter and a controlled XY->MS->XY matrix. It can be active only during stereo-signal processing.

This algorithm has been designed to enable the correction of the stereophony without introducing significant distortion. The block diagram of this algorithm is presented below.



In order to ensure the phase compliance of the signal with the range of treble frequencies, the main emphasis filter has been designed as a finite impulse-response filter (FIR filter), which ensures a linear phase. The first element of the algorithm is the emphasis filter with a linear phase algorithm. The picture below shows the filter operation dependent on the setting of the filter slope parameter.

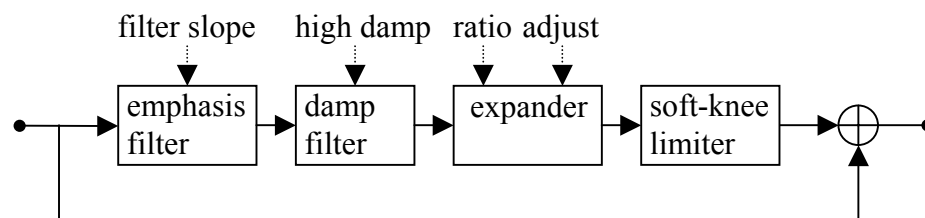


Another element of the algorithm is the XY->MS->XY matrix, in which the correction of the side and middle signal content takes place, and the signal processed is mixed again with the signal which is not processed.

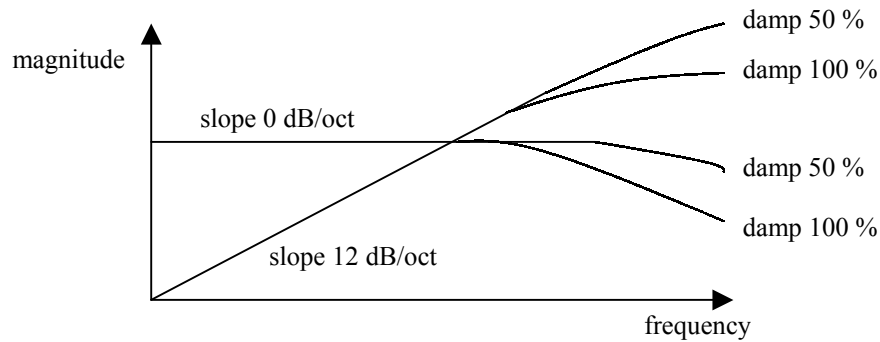
Transients section

This section is designed for enhancing transients which have been flattened by poor-quality analog equipment. The operation of this section is based on the high-frequency compander. Due to this algorithm, it is possible to revitalise tracks whose details have been softened, as well as to improve definition in the medium and high-frequency range, without an increase in the noise content.

This algorithm has been designed so as to enable an increase in the content of high harmonics in the signal based on the existing signal but without increasing the harmonic content. That is why this algorithm is useful in processing both single tracks and whole mixes. The structure of this algorithm is shown in the diagram.



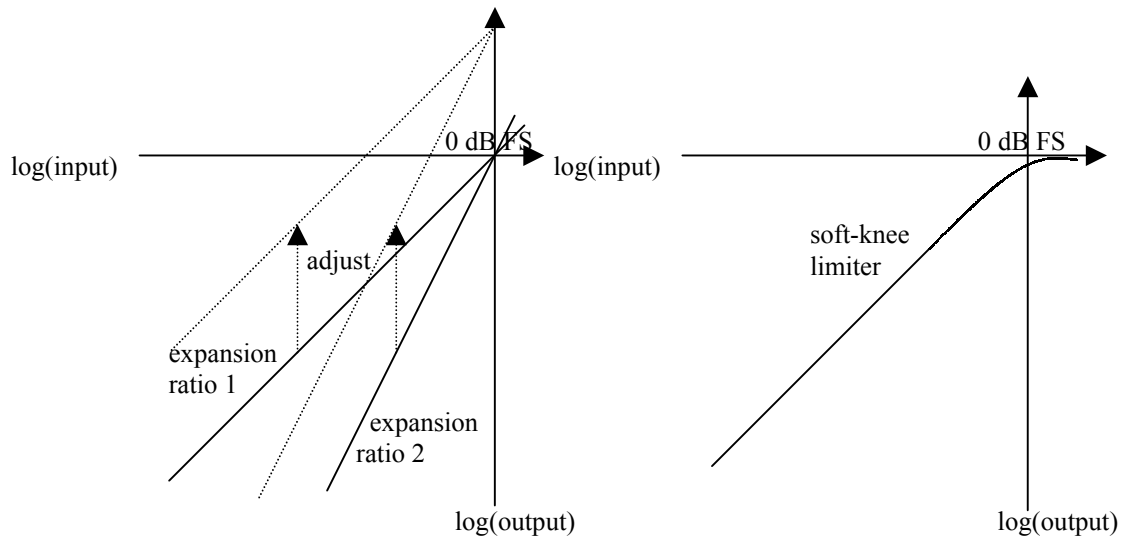
In order to ensure the phase compliance of the signal with the range of treble frequencies, the main emphasis filter has been designed as a finite impulse-response filter (FIR filter), which ensures a linear phase. In order to avoid over-brightening of extremely high frequencies, this circuit has been equipped with a low-pass filter with a gentle characteristic of 6 dB/oct. The picture below shows the filtration curves dependent on the setting of the filter slope and damping parameter.



After the signal enters the equalizer within a range of treble frequencies, it is processed using the expansion algorithm. The time parameters of the expander and compressor are chosen automatically in such a way as to ensure the correct and rapid reaction of the system according to the previously set range of the equalizer. Inside the expansion process, the regulation of the 'adjust' parameter takes place, which is directly responsible for the algorithm's operational depth.

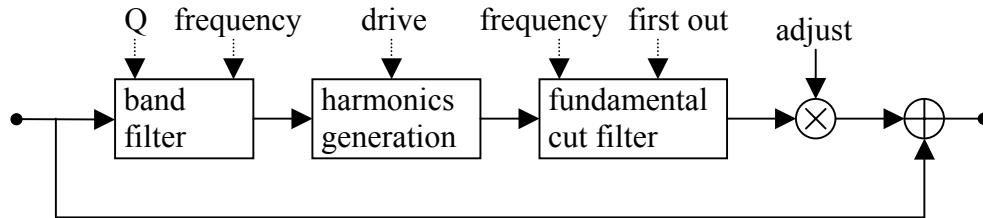
In order to prevent the occurrence of over-emphasized transients as a result of expansion, this algorithm has been equipped with a soft-knee compressor/limiter, situated directly before the point at which the processed signal and input signal are mixed.

The diagram below shows the expansion and compression curves in the processing path of this algorithm.



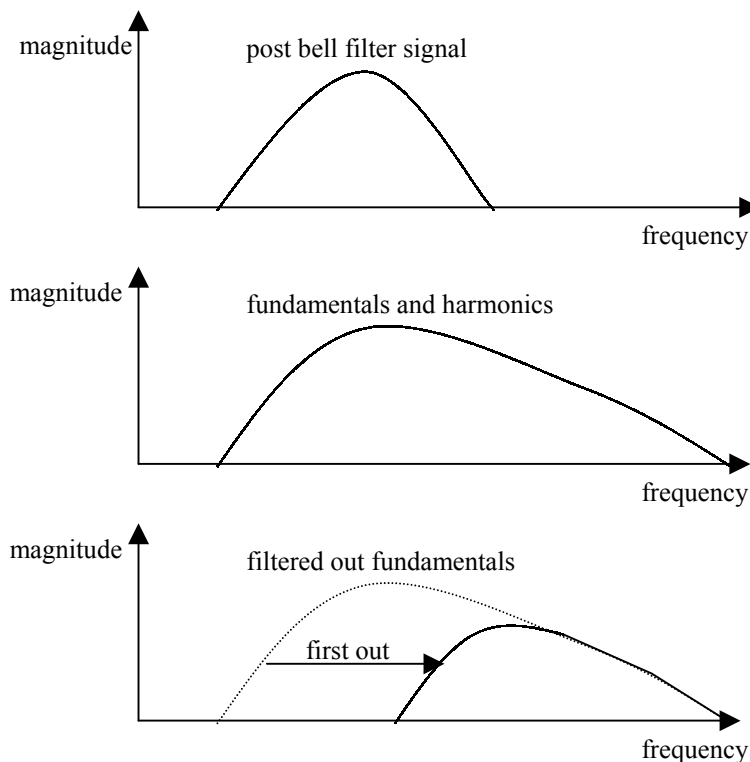
Harmonics section

This section is designed for generating additional harmonics, for improving clarity and definition, or adding presence to a track. This algorithm has been equipped with a wide-range tuneable bell-type filter. The filtered signal is sent to the harmonics generator and then to the fundamental component-removing filter. The harmonics generator generates odd and even harmonics. See block diagram below for details.



This algorithm resembles typical circuits of the exciter type which add a presence to the signal through correction combined with harmonics generation.

A bell-type filter, which selects the frequency range for further processing in the signal path of the algorithm, is controlled precisely within a wide range by the 'frequency' and 'Q' parameters. The 'drive' parameter is responsible for the depth of operation of the harmonic distortion generator, and the 'first out' parameter enables the setting of the relative frequency of the fundamental-component cut-off. These frequencies depend on the setting of the frequency and Q of the bell-type filter. The graph below shows the way those filters work.



The output signal of the algorithm is mixed with the input signal by the Adjust parameter.

Like other plug-ins from the MixPack 1.5 series, the PSP MixTreble's 'Mix' control can be used for mixing the input signal with the processed signal in the desired proportions. This function allows for quick and comfortable setting of the optimum processing depth. This regulation is before 'output' control of the plug-in and does not influence directly to the operation of soft-clipping algorithm.