Pro Audio DSP DSM V3

Plugin Manual

Developed by Pro Audio DSP and Brainworx Audio distributed by Plugin Alliance

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Pro Audio DSP DSM V3

Plugin Manual

The Dynamic Spectrum Mapper (DSM) plugin is intended to provide multi-dimensional dynamic control over both the spectral response and dynamic characteristics of audio programme, in order to bring a whole new dimension of facility and artistic ability to the sound engineer. The use of novel processing and programme capture algorithms provides the ability to achieve exemplary results in common tasks such as programme compression and enhancement, vocal expression and instrument characterisation very quickly. The ability to capture, modify and re-apply the spectral characteristics of programme is a powerful creative tool in all stages of the production cycle, from recording and mixing to mastering stages.

New in DSM V3

• All new Expand mode.

• Fully active spectrum sidechain input with source selection.

• Active Drag handles on Spectrum graph.

• Limiter and DSM dynamic processing can now be used independently.

• Improved signal flow for greater convenience.

• More convenient setup and curve store and restore management.

• Scalable GUI from 50% to 150%.

• Allows import of DSM V2 capture curves

Pro Audio DSP DSM V3

Plugin Manual

Principle of operation

The DSM is a dynamic compression and expander application. The basic architecture of the plugin is similar in concept to a multi-band compressor and expander, except that the method of band splitting produces far greater frequency resolution in the higher mid and high frequency ranges.

This high frequency spectrum resolution allows a more natural and musical compression, it also provides a high degree of functionality and artistic modification of the programme character. Since the process produces different artefacts to conventional band splitting dynamics applications, a new range of programme character can be achieved, such as tape saturation effects and the softening of high frequency content.

The signal is split into multiple bands using an FFT based process. An adaptive dynamic gain control process operates on the effective power represented within each band, before recombination back into the output signal domain.

Manipulation of the threshold curve changes the threshold target level for each band. This allows the frequency content to be modified based on dynamic changes in the signal over both time and rate. Because the dynamic gain control process includes frequency domain information captured from the original programme (as well as user-defined target responses), it provides a large degree of novel artistic functionality, which includes:

Pro Audio DSP DSM V3

Plugin Manual

In Compression mode

- Enhance the perceived volume and presence of mixes.
- Compress even the most difficult programme with increased accuracy and musicality.
- Characterise and personalise output in completely new ways.
- Copy and equalise sonic impressions between dissimilar sounding tracks.
- Process, enhance and 'tame' vocal tracks with unprecedented accuracy and speed.
- Dramatically enhance and increase the intelligibility of dialogue for media post production.
- Model and characterize instrument sounds to achieve highly creative artistic impressions.
- Create unique dynamic effects in both level and frequency domains to produce exciting new artistic effects.
- Control the dynamics of programme with content from another channel, via the sidechain input.

In Expansion mode

- Increase overall dynamic range.
- Accentuate the level and spectrum content in parts of the programme.
- Accentuate or gate programme with level and spectrum content from another channel, via the sidechain input.

Pro Audio DSP DSM V3

Plugin Manual

Parameter Description

1 Graph

The graphical display area plots level against frequency and contains two curves.

The blue curves (stereo) or yellow curve (linked mode) displays the Active Spectrum of the input signal at all times and is live during programme.

The white curve displays the Threshold Spectrum of the captured (or manually modified) spectrum. This sets the threshold of compression in each band.

The white active threshold spectrum is obtained by operating the Capture button. The button can be operated momentarily to capture a short transient spectrum, or if held down it will average the spectrum over time in order to capture a longer-term character spectrum of the programme. Note. In order to accurately match the input signal, the Capture function acquires both the level and frequency spectrum of the programme. Therefore the position of the white threshold curve on the graph display (and the resulting compression thresholds) will initially depend entirely on the level of the input programme. Quieter and lower level programme will result in lower captured threshold levels.

Note: The active threshold spectrum is saved along with all other parameters in the saved plugin settings.

Therefore captured responses can be saved and re-used on other tracks and projects.

Pro Audio DSP DSM V3

Plugin Manual

In Compression mode, dynamic gain change can occur anywhere on the graph where the blue curves (stereo) or yellow curve (link Mode on) is higher than the white threshold curve. Where the blue or yellow curve remains beneath the white curve the signal at those frequencies will remain unchanged.

In Expand mode, dynamic gain change can occur anywhere on the graph where the blue curves (stereo) or yellow curve (link Mode on) is lower than the white threshold curve. Where the blue or yellow curve remains above the white curve the signal at those frequencies will remain unchanged.

1 Input Meter

The Input meter shows signal level before any processing happens. The values are displayed in dBFS. Click on the peak indicators in order to reset them.

2 Output Meter

The Output meter shows signal level after all processing. The values are displayed in dBFS. Click on the peak indicators in order to reset them.

3 Process

Enables/Disables compression and expansion processing. Can be used for comparison without generating any audio artefacts.

4 Mode

Switches between Expander and Compressor modes.
Please note that the DSM can only operate in one mode at a time. I.e. it cannot act as a compressor and expander simultaneously.

Pro Audio DSP DSM V3

Plugin Manual

1 Freeze

The Freeze function stops dynamic activity and freezes the response to that which was active at the moment of operation. This arrests any further gain compression and permanently imposes the response of the active spectrum onto the programme – somewhat like a static equaliser.

2 Capture

Captures the frequency curve and level information of your programme, which is applied thereafter.

Press once for a spot capture, or hold the button down for an average capture over time.

Capture curves can be saved and loaded independently of presets.

3 Load

Load a previously saved capture curve from file.

Note: If you own DSM V2 you can also import V2 capture curves by simply loading them.

4 Save

Save the recorded capture curve as a file.

5 Link

The Link button ties left and right channels together for stereo operation. The curve is coloured yellow to indicate stereo linking. If the Link is deselected, left and right channels will operate independently.

During independent operation the yellow curve is replaced by two blue shaded curves representing the active spectrums of both left (dark blue) and right (light blue) channels.

Pro Audio DSP DSM V3

Plugin Manual

1 **Log**

Toggles between linear and logarithmic display of the frequency response.

2 **Limiter**

Enables the Limiter function, which suppresses sample value overloads. It achieves this by reducing the peak levels of the signal above -1.0dBfs without significant loss of loudness. It also prevents hard clipping up to +6dB above 0dBfs and (depending on the programme material) allows higher levels to be developed within the DSM section without clipping, which can significantly enhance the loudness and presence of the programme.

When the Limiter is engaged, the Red overload section of the Output Meters changes from Red to Yellow to indicate the degree of limiting. While the Limiter is engaged no output greater than 0dBfs is possible – so the over indication on the meters does not light.

Please note that the Limiter is the last process in the signal chain and acts separately from the spectrum dynamic section. I.e. The Limiter functions when the Limiter button is active, even if the Process button is set to out.

3 **Dither**

The 16 Bit Dither button toggles from 24bit to 16bit target media when selected.

Pro Audio DSP DSM V3

Plugin Manual

Detector

Controls the signals entering the dynamics processing from the direct and sidechain sources.

Detector Modes

1 Direct

This feeds signal from the main input to the dynamic control. This is the normal mode for direct track operation.

2 Sidechain

This feeds signal from the sidechain to the dynamic control. (please check your host manual in case you are unfamiliar with the routing of sidechain input signals).

3 Max

This responds to both input and sidechain signals and operates on the signal which has the maximum signal level within the active frequency bands.

4 Listen

Listen exclusively to the sidechain signal. Make sure you disable this when done.

5 Sidechain Gain

Controls the gain for the sidechain signal. Use this to adjust your sidechain signal to obtain optimum signal level.

Pro Audio DSP DSM V3

Plugin Manual

Dynamics Section

The dynamics section contains controls for the dynamic activity of compression and expansion. They are presented in a fashion which is familiar to users of conventional parametric dynamics sections – and behave in exactly the same manner.

1 Threshold

The Threshold control changes the level of the Active Threshold curve in relation to the captured spectrum. The white active threshold curve position changes vertically on the graph to reflect this.

When the Threshold control is set to 0dB the active thresholds correspond to the active levels of the original captured (or manually modified) spectrum.

If the Threshold control is adjusted to lower (or higher) levels, the active thresholds in all bands will vary accordingly. An extended positive threshold range beyond 0dB is provided so that Threshold Spectrums captured at lower programme levels can be raised to useable levels.

In compression mode, gain reduction will occur for any levels in the spectrum range that are larger than the Spectrum Threshold curve.

2 Ratio

When in Compressor Mode, Ratio controls the ratio of gain compression for all frequency bands.

When in Expander Mode, controls the ratio of gain expansion for all frequency bands.

Pro Audio DSP DSM V3

Plugin Manual

1 Gain

The Gain control is provided as an independent manual gain make-up function to compensate for level reduction due to compression.

2 Attack / Release

In compression mode the Attack and Release controls modify the rate of gain reduction and recovery common to all frequency bands.

In expansion mode the Attack and Release controls modify the rate of gain increase and recovery common to all frequency bands.

3 Knee

The Knee control controls a 'soft' compression ratio function and is expressed by the number of decibels below the active threshold that the soft knee will start pre-compressing, in advance of the active threshold being reached.

Pro Audio DSP DSM V3

Plugin Manual

Timing

Normally all time constant parameters within the dynamics section are common to all bands in the spectrum. The timing profile controls provide a method to modify timing parameters in relation to frequency range, in order to produce more sympathetic action depending on programme type, or to produce sound effects.

1 Timing LF

In compression mode the Timing LF (low frequency) progressively increases the attack time for lower frequency signals in relation to higher frequency signals. This slows down the attack for low frequency compression events within the spectrum.

In expansion mode the Timing LF control increases the release time for lower frequencies. This slows down the release for low frequency expansion events within the spectrum.

2 Timing HF

In both the compression and expansion modes, the Timing HF (high frequency) progressively decreases the release time for high frequency signals in relation to lower frequencies. This speeds up the gain recovery for high frequency compression events.

Pro Audio DSP DSM V3

Plugin Manual

Master

This section is provided to allow the control of the level of the dynamically processed signal and the blending of it with the input signal.

Please note that the Limiter section follows this whole section in the signal path and is therefore not defeated by it.

1 Master Dry/Wet

Adjust the mix between dry (input) and dynamically processed signal – prior to the Limiter section. This allows you to blend the dynamically processed signal with the input signal.

2 Master Output Gain

Adjust the gain of the dynamically processed signal – prior to the Limiter section. This helps to compensate for level changes.

Pro Audio DSP DSM V3

Plugin Manual

Frequency Bands

There are 3 frequency bands red, green and blue. All frequency bands have the same features and cover the whole frequency spectrum (100Hz – 20kHz).

The purpose of these control sections is to allow modification of the Spectrum threshold after capture.

They are presented as familiar 'EQ style' controls for convenience. But they do not actually EQ the signal itself.

When the bands are active, full control of the parameters can also be accomplished by drag handles which appear directly on the Spectrum threshold itself.

Note: The controls neither equalise the signal itself, nor act as sidechain EQ. They only affect the thresholds within the dynamic bands.

❶ Frequency Band Enable/Disable

Enable or disable processing for this Frequency Band. This can be used for comparison purposes.

The response modifications produced by the parametric threshold control are added non-destructively to the existing active threshold curve and parameters are stored separately, so that a set up can be restored and modified further without loss of the originally captured spectrum.

❷ Frequency

Controls the centre frequency of the parametric threshold.

Pro Audio DSP DSM V3

Plugin Manual

1 Threshold

Adjust the Threshold for this frequency band.

2 Q Factor

Adjust the Q-factor for the corresponding frequency band.

Pro Audio DSP DSM V3

Plugin Manual

Presets

Extensive tools are provided to enable the storing, restoring and management of presets spectrum capture curves.

A preset always contains settings for all four banks (A/B/C/D). Preset operations like saving and loading will affect all four banks.

1 Preset Dropdown Menu

The dropdown menu lists all your presets by category. Categories can be added when creating a new preset. A category will be listed as long as there are presets within this category.

The Favorites category is always listed and holds all presets which you have marked as favorite.

2 Save/Overwrite Preset

Clicking on this button allows you to choose between overwriting or saving the current preset as a new preset. Saving a new preset opens a dialog giving you the possibility to fill in the preset name and category. You can choose from a list of existing categories or create a new one by starting to type.

Please Note: It is possible to overwrite factory presets.

3 Delete Preset

Deletes the currently active preset and banks (A/B/C/D).

Please Note: It is possible to delete factory presets. Reinstalling the plugin will reinstall deleted factory presets without overwriting changed factory presets.

Pro Audio DSP DSM V3

Plugin Manual

1 Mark as Favourite

Adds or removes a presets to your favorites. There is a separate category in the preset dropdown menu allowing to conveniently browse your favourites.

2 Up/Down

Step through the preset list. Jumps to the previous/next category when the first/last preset is currently selected.

Top Toolbar

3 UI Scaling

Adapt the graphical user interface to the size and resolution of your screen. The plugin will check if the user interface will fit your screen before enlarging it. So if you accidentally chose a zoom factor which is too big for your current settings, the plugin will automatically stay at the maximum possible zoom factor.

4 Undo/Redo

You can undo and redo changes you made to the controls of the DSM V3 plugin at any time. The Undo / Redo will work for as many as 32 steps. This makes experimenting and tweaking knobs easy. If you don't like what you did... just undo it.

Pro Audio DSP DSM V3

Plugin Manual

1 Settings (A/B/C/D)

The Plugin offers four internal settings (A/B/C/D) which will be stored with every preset. So, one preset can contain up to four settings. You may use similar settings with more or less compression in one setup / preset. Now, the SETTINGS can be automated in your DAW! This way is possible to use different sounds for your lead vocals or drums in various sections of the song. Automate the A/B/C/D settings, and you can still tweak knobs of the individual settings without overriding multiple parameters in your DAW, which would be time-consuming.

2 Copy / Paste

To set up variations of similar sounds you don't have to dial in the settings several times. Let's say you like your setting A and want to use the same sound, just with less compression, as setting B.

• Simply press Copy while you are in setting A.
• Switch to setting B by pressing 'B' in the settings section.
• Press PASTE, now setting B is identical to setting A.
• Reduce the compression on the B setting.

Now you can switch between A & B and decide which one sounds best or automate different settings for various sections of your session.

3 Reset Settings

Resets the currently selected setting (A/B/C/D) to the default parameter state.

Pro Audio DSP DSM V3

Plugin Manual

Using the DSM in Compression mode

With an understanding of the general architecture of the DSM, functionality should be quite straightforward and there are many ways to achieve useful results. The following basic procedures are suggested as operational routines to deal with common situations and illustrate functionality.

Setting the Capture response

The DSM differs from conventional dynamic compressors in that it makes use of the original spectral character of the programme. Therefore it is important to establish the threshold spectrum initially.

The best way to hear the capture is to set a gain limiting situation by setting the **Threshold control** to –24dB, the **Gain** to +24dB, **Ratio** to 100:1, put **Attack** to around 10mS and **Release** to 30secs, leaving all other controls at minimum.

This will cause the compressor to gain limit the signal to the captured response.

The '**start**' set-up is included for convenience and provides these settings.

Operating the **Capture** function will then sample the spectral parameters of the input programme. It is often best to do the capture at a place in the programme where it is fully under way – most often this equates to the loudest sections.

19

Pro Audio DSP DSM V3

Plugin Manual

At this point the programme will be effectively gain limited to the response of the captured spectrum. This can be confirmed by toggling the **Process** button to turn the processing on and off, where the processed sound should closely resemble the original programme over the passage where the capture was taken.

You now have a compression that is removing all dynamic range for signal content above the **Threshold Spectrum** and forcing it to take on the spectrum and level of your captured response.

This is a very good starting point that we will return to throughout operational description.

Setting Dynamic parameters for compression

The **Threshold control** can be used to change the active level of the captured threshold spectrum to compress to a higher or lower output target level, or place the threshold spectrum at higher levels in order to impose the captured compressor response and dynamic control only on to louder parts of the programme. For reference, live changes in the levels of the active spectrum of the music are reflected in the blue curves (stereo) or yellow curve (linked mode) of the graph.

The **Ratio control** can be used in a conventional way to restore the required amount of dynamic range for signal within the compression range and the **Knee control** can be used to soften the transition in and out of the compressed level ranges. The **Gain control** can be used to compensate for level changes caused by threshold and ratio adjustments.

Pro Audio DSP DSM V3

Plugin Manual

The timing controls act in the same way as a conventional compressor. Increasing the **Attack time** will slow response to peak events, reducing the **Decay time** will speed up gain recovery after compression events.

The **LF** can be employed to slow down compression of low frequency events to allow greater perceived LF energy in the sound.

The **HF** may be used to speed up the recovery of high frequencies to produce the effect of increasing overall HF energy, 'air' and 'fill' to the sound.

Note, since the capture function employs the same processing archi-tecture as the signal control, *compression gain variance due to timing control changes can be compensated for by re-capturing the spectrum with the new active settings.* This is particularly useful when setting timing controls for effects. Very wide time constant settings can be accommodated and experimented with in this way.

Due to the large numbers of effective bands being compressed and the included 'weighting' from the original spectrum of the programme, a very sympathetic and credible compression can be obtained over a very wide dynamic range. In particular, percussive events at higher frequencies do not modify lower frequency ranges as in conventional compressors and the existence of the original spectrum maintains a credible sonic signature, even during large amounts of gain compression.

Pro Audio DSP DSM V3

Plugin Manual

Character modification

The frequency spectrum character of a source track or sound can be applied to another simply by capturing the source and playing the target track through the plugin.

Going back to our starting limiting set up in *[Setting the Capture response]*, operating a capture on the source track will force the output to follow the captured response, If the target track is played through this (without re-capturing) it will take on many of the spectral characteristics of the source sound.

The dynamic controls may be used like a conventional compressor to achieve the desired dynamic range characteristics

If dynamic compression activity is unwanted, the **Freeze** function can be used to suspend compression and the plugin will operate like an EQ which has the source character pre-loaded. Slower attack and release settings are best for this function, as the slower response time will facilitate a more representative frequency response at the moment the gains are frozen.

Novel **Spectrum Thresholds** can be constructed by modifying an existing one or building completely from scratch using the **Parametric Threshold** section. Set-ups with pink and white frequency spectrums are provided as useful starting points if building spectrums from scratch.

Pro Audio DSP DSM V3

Plugin Manual

De-essing

This is a process of simply capturing the vocal response during a period where 'essing' does not occur and re-applying it to the vocal track at the appropriate threshold levels. Some included set-ups illustrate this.

Going back to the original limiting situation we had set up in *[Setting the Capture response]*, setting both the **Threshold** and **Gain** to 0dB will ensure that any capture will be calibrated to unity at the incoming signal level.

If a capture is performed during a period where the singer is not 'essing' and this is applied to the whole track, louder essing events will breach the **Threshold Spectrum** within the frequencies they occur and those parts of the spectrum will be forced back to the captured response.

Since the captured response is a valid spectrum of the singer's original voice, the quality of the de-essing can be very high.

If desired, the **Ratio** and **Knee** functions can be adjusted to achieve a progressive application of the target response and the **Threshold** can be adjusted so that the effect commences at the appropriate levels.

Alternatively (or in combination with the above) one can visually identify essing events in the graph area and a custom threshold spectrum can be constructed using the **Parametric Threshold** section to build an appropriate response with a 'dip' in the offending region. This can then be applied at the appropriate levels to reduce these events.

Pro Audio DSP DSM V3
Plugin Manual

Faster **Attack** and **Release** times can often be used effectively because the action is restricted only to high frequency events.

Slower **Attack** times can be effective at retaining the sibilance of the voice whilst still suppressing unwanted longer term 'essing'.

Vocal enhancement

Vocal enhancement and intelligibility improvement are conceptually similar to the above; except that the character and dynamics of the voice are deliberately modified under a range of levels, programme and timing conditions to achieve the desired result.

The whole voice may be compressed to a captured spectrum of itself, to achieve maximum consistency and penetration whilst retaining vocal character. Or it may be compressed with a **Spectrum capture** of some other vocal parts or tracks in order to achieve a higher degree of consistency between takes or to emulate aspects of other tracks.

It can be made to 'morph' from one character into another by either applying a previously captured spectrum or creating a custom **Threshold Spectrum** in the **Parametric Threshold** section, setting the threshold at some intermediate level and applying the appropriate compression ratio and timing parameters.

Setting lower ratio values will cause the sound to take on the character of the **Threshold Spectrum** progressively as the voice gets louder.

Pro Audio DSP DSM V3

Plugin Manual

By manipulating the **Attack** and **Release** controls it is possible to cause the voice to blend towards the **Threshold Spectrum** over a period of time without crushing the sibilance. This can be useful for preferentially moderating sustained vocal phrases, particularly when they can often have a tendency to become harsh in many vocal styles.

Because the compression is aiming for a credible spectrum of the original voice, explosives and other unwanted vocal sounds are effectively tamed without a damaging loss of character to the overall vocal performance.

Loudness and presence enhancement

Using the **Limit** function in combination with the **DSM** level control and compression can result in significantly greater loudness and presence for both single tracks and entire mixed programme. Because the **Limit** function allows higher modulation levels without clipping, it is possible to pass signals that would otherwise cause sample value overloads.

This can be achieved after setting the appropriate **Threshold Spectrum**, **Attack** and **Release**, by increasing the overall threshold or gain a few dB in order to progressively overdrive the **Limit** processing. Tight control of the levels by the DSM can avoid gross distortion from the limiting section.

Whilst it is possible to use a **Threshold Spectrum** captured from the programme itself, it is also possible to achieve interesting results by producing a modified threshold spectrum that delivers greater gain in the mid ranges (where we are most sensitive) and allowing the DSM to impose this response onto the louder or percussive parts of the programme.

Pro Audio DSP DSM V3

Plugin Manual

To achieve this, higher overall threshold settings can be used to apply a modified (artificial) threshold spectrum only to the top few dB of the dynamic range.

Set-ups named as 'louder' are included as an illustration of this concept. The starting point for most of these was the 'pink' set up, from which the spectrums were further modified using the **Parametric Threshold** section.

Note: For percussive programme, greater overall punch may be achieved by setting slower attack times and allowing the limiter to handle the high level transients.

Note: For maximum loudness, Knee function settings should be miminised to avoid premature compression.

Sound effects

The novel architecture of the plugin produces a different set of artefacts from conventional compression and this can be used for a range of sound effects. Because the DSM can capture and aim for specific responses, novel effects may be generated if used in conjunction to other processes such as equalisation. Some examples are provided below.

High frequency content softening

Because the compression occurs within many bands simultaneously, very fast attack and release times can be used over relatively small level ranges, particularly in the higher frequency regions.

Pro Audio DSP DSM V3

Plugin Manual

By applying a suitable Threshold spectrum that preferentially compresses the HF content, a useful softening of the high frequency content can be achieved which is reminiscent of tape saturation effects. Some included set-ups illustrate this effect.

High frequency content enhancement

By applying significant compression at lower thresholds and ratio ranges, then preferentially speeding up the recovery of the HF compression using the **HF** control, a pleasing increase in HF detail and presence can be produced, which is reminiscent of a style processing known as 'aural enhancement'.

Unlike the application of simple equalisation, because the HF levels remain under control by the compression, no extra harshness is produced.

This can be particularly effective when applied to acoustic stringed instruments, woodwinds and flutes to accentuate the high frequency overtones.

Low frequency enhancements

Increasing the **LF** control will preferentially slow down the attack speed for low frequency programme content. This is useful as a way of increasing the perceived warmth of the sound and preventing short-term percussive resonance to be cut short excessively.

By producing a custom **Threshold Spectrum** that rolls off in the lower frequencies and placing it at a suitably high level, it is possible to cause the LF to be moderated at high levels. This is useful for reducing LF over

Pro Audio DSP DSM V3

Plugin Manual

loading when aiming for the maximum possible loudness in programme with significant dynamic range.

By applying equalisation prior to the **DSM**, it is possible to pre-boost certain parts of the programme and allow the DSM to re-normalise it at higher levels, to effectively 'undo' the equalisation. Effects such as low-level sibilance enhancement, low level warming and other dynamic frequency effects can be produced this way.

Using the Sidechain in compression mode

The sidechain input can be used to drive the compression of the DSM from signal sourced from another channel or track.

Since the DSM sidechain is fully operational in the spectral range, it can produce a far greater artistic capability, compared to ordinary compressors which operate only on peak full range signal level. This produces a whole new dimension in artistic capability.

For instance, if you put the DSM in a track (or buss) and feed the DSM **Sidechain** with signal from the kick drum, the spectrum of the kick drum will become dynamically attenuated in the target track, each time the kick drum is played. But the rest of the frequency range will remain signifi-cantly unaffected. The same will be the case for all manner of combina-tions of instruments and sounds.

Since you can also capture **Spectrum thresholds** on the fly directly from the sidechain sources via the sidechain, there are a multitude of possibi-lities in real mixing situations which can be of novel artistic interest.

Pro Audio DSP DSM V3

Plugin Manual

The **Max** setting on the sidechain input section allows the DSM to respond to whichever is the higher signal in any frequency range, which means you can blend the sidechain effect with the direct compression.

The **S/C gain control** allows you to set the appropriate contribution of the sidechain input to obtain the optimum effect.

Using the DSM in Expansion mode

The DSM expansion operates in a mode which is known as **Downward Expansion**.

This means that (unlike a compressor, which dynamically reduces gain for signals **above** the threshold), the **Expander** instead dynamically **reduces** the gain of signals **below** the threshold. I.e. the lower the signal level is, the greater it is reduced further.

In this way the dynamic range of the programme is increased – or 'stretched' in other words.

The amount of dynamic reduction in the gain for signal below the threshold is controlled by the **Ratio** control.

In expand mode the **Ratio** control changes scales to a range of ratio setting from 1:1 at the bottom – (i.e. no expansion), to 4:1 at maximum – (i.e. 4 times expansion). This means that increasing the **Ratio** control will cause a proportionally greater reduction in gain for signals below the threshold, up to a maximum of 4 times reduction in the level of the original signal (in decibels).

Pro Audio DSP DSM V3

Plugin Manual

For convenience, the **Ratio** control settings for compress and expand are treated and saved separately, so you can switch between them without loss of settings.

Apart from this, all other controls in the dynamic section operate as they do in compress mode.

Increasing overall dynamic range

To increase the overall dynamic range of the music, the aim is to set the **Threshold** somewhat higher than the quieter parts of the music, so that the signal below the threshold is dynamically quieter than it was in the original music signal.

Everything above the threshold will remain at the level of the original signal.

The **Ratio** varies the amount of the expansion effect – and the **Attack** and **Release** controls can be set to optimally match the timing of the music itself.

Pro Audio DSP DSM V3

Plugin Manual

Spectrally enhanced expansion

Since the **Expand mode** has full spectral capabilities, portions of the frequency range which are loudest will become more dominant for signals below the threshold, since they will approach the threshold more closely.

However, when all bands are above the threshold, the balance is the same as the original.

This means that artistic effects can be created in your musical signal, to enhance the level range where signals are quieter. For instance, it can accentuate low frequency bass parts, or conversely accentuate high frequency cymbal crash parts, depending on the Threshold capture spectrum.

Pro Audio DSP DSM V3

Plugin Manual

Using the Sidechain in expansion mode

The **sidechain** input functions in expansion mode exactly as it does in compression mode.

It enables the control of the expansion gain from signal derived from another track or source.

Since the sidechain has full spectral characteristics, it can be used to dynamically impress the response of the sidechain source track on to the target track, where the sidechain signal spectrum will dynamically increase the gain within the target track.

For example, if you put the DSM in a track (or buss) and feed the DSM sidechain with signal from another instrument track, the spectrum of the other instrument will become dynamically increased in level within the target track, each time the other instrument is played. However the rest of the frequency range will remain significantly unaffected. The same will be the case for all manner of combinations of instruments and sounds.

It is possible to use a capture from the main signal, from the sidechain, or indeed from any other saved spectrum capture, to produce a range of artistic effects.

Pro Audio DSP DSM V3

Plugin Manual

Peak levels and Limit function

Because the DSM modifies the frequency response, and therefore changes the waveform shape of complex signals, it is possible that larger peak levels will result. As in the case of simple equalisation, this is true even when reduction is applied at certain frequencies. Therefore if maximum peak modulation is required, the **Limit** function should be employed to suppress sample value overloads in the output programme.

Relative frequency distribution

The DSM uses FFT methods to produce the active frequency bands. This means that the density of effective bands naturally increases towards the higher frequency ranges, producing the greatest frequency resolving ability at high frequencies.

Whilst this provides a high degree of control in the higher mid and HF frequency ranges most identified with overall programme character, the natural reduction of active bands in the LF region may make it unsuitable for some classes of sound effects that focus on narrow band low frequencies within the programme.

Latency

The total procesing delay is the same for all formats:

44.1KHz & 48Khz: 49 samples

88.2KHz & 96KHz: 98 samples

176.4KHz & 192KHz: 196 samples