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Limiters: Theory & Practice Part 1

How they function and how to use them correctly

Introduction

All NST processors[1] feature two stages of limiter protection on all outputs – an RMS limiter designed to control sustained maximum output level (and so manage speaker thermal stress), and a look-ahead clip limiter, designed to prevent over-excursion (and so reduce mechanical failure).

RMS Limiter

High performance limiters are provided for each output with control over attack time, release time and threshold parameters. This level of control allows the user to balance the required subjective quality of the limiter against the driver protection requirements. Be aware that a limiter set incorrectly can sound worse than no limiter at all!

In particular, as with all limiters, using too fast an attack or release time for the type of signal in the pass-band will result in excessive low frequency distortion. There is provision, in the output sections of each processor's configuration within D-Net, to set automatic limiter time constant on an output by output basis. Use this option if you are unsure how to set the time constants manually. We recommend the use of the automatic setting.

In this mode the time constants will be automatically set from the corresponding channel's High-Pass filter frequency according to the table below.

High Pass Filter	Auto Attack Time	Release Time
<10Hz – 31Hz	45mS	x16 (720mS)
31Hz – 63Hz	16mS	x16 (256mS)
63Hz – 125Hz	8mS	x16 (128mS)
125Hz – 250Hz	4mS	x16 (64mS)
250Hz – 500Hz	2mS	x16 (32mS)
500Hz - 1kHz	1mS	x16 (16mS)
1kHz – 2kHz	0.5mS	x16 (8mS)
2kHz – >32kHz	0.3mS	x16 (4mS)

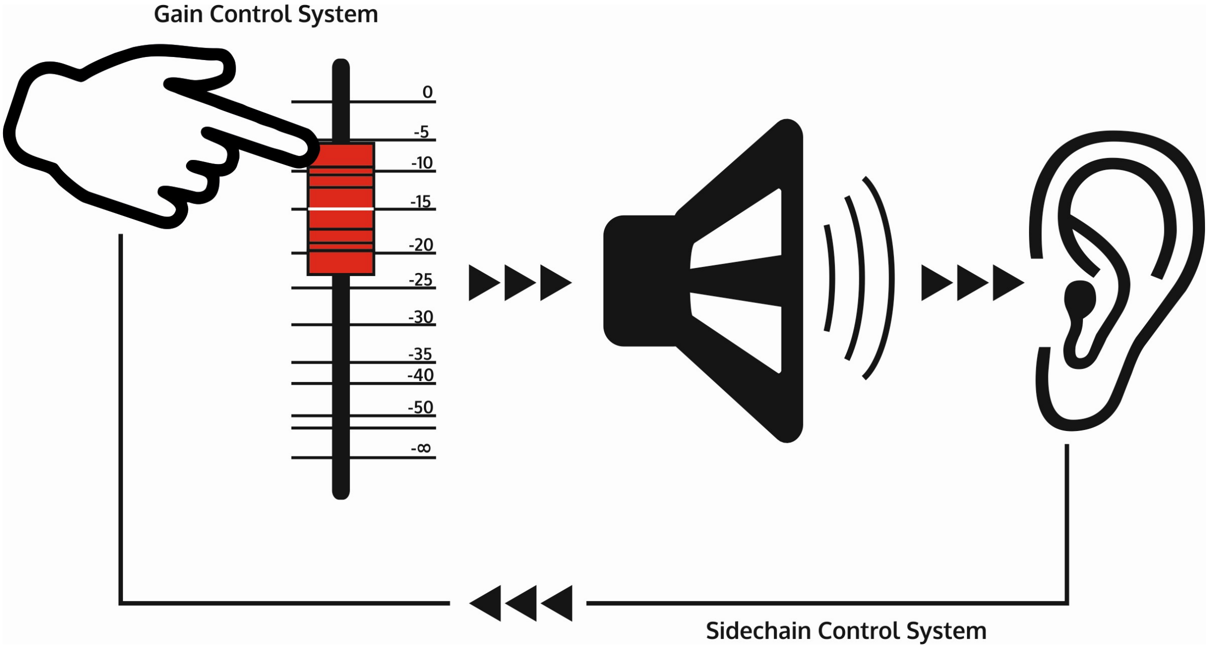
Peak Limiter

The main limitation with traditional dynamics control is the inability of the processing to react truly instantaneously to the signal. One of the most significant advantages of digital signal processing over analogue is the ability to delay the audio signal precisely and without extensive complex hardware. The entire domain of digital signal processing is based around the combination of delaying, multiplying, and accumulating numbers (representing samples of audio) to implement all the filters and dynamics processing we have come to expect today.

[1] Original D48 adds clip limiter functionality on outputs in 48kHz mode.

In the case of dynamics processing, being able to delay a signal allows the processor module to delay the main signal in relation to the sidechain (the signal being monitored relative to the threshold), so that it can compensate for peaks prior to the arrival of the main signal.

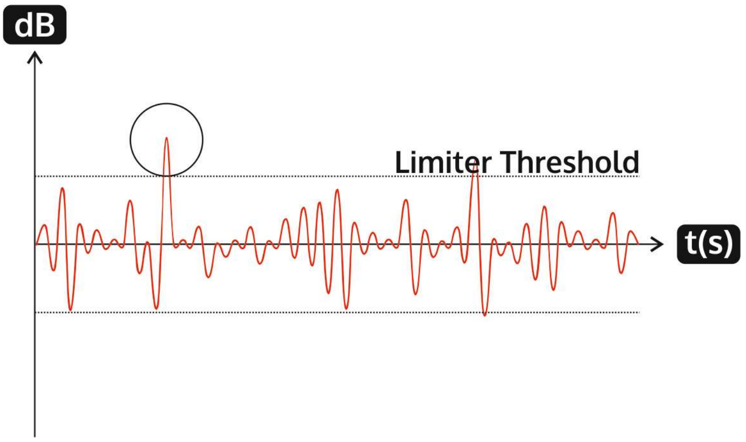
Consider the situation of a monitor engineer listening to a band perform. Having no access to dynamics processors, he has had to resort to manually 'riding the faders' in an attempt to keep control of the levels. Should the level of one of the channels on his desk reach an unacceptably high level, he will turn it down appropriately.



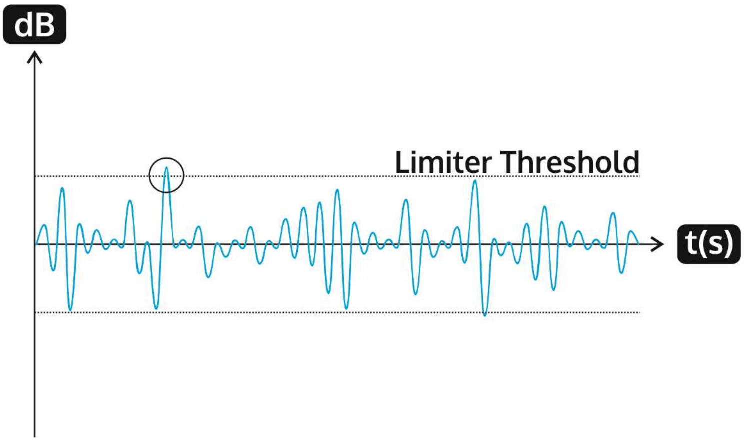
The main signal path is fed through the monitor desk and the gain controlled by adjusting the fader. The sidechain is formed by the feedback path between the engineer's ears checking the level and his brain instructing his hand to turn the fader down if the volume goes over the threshold he has chosen.

In this case, the delay between the signal actually going over the threshold, the engineer registering the situation, and then turning the signal down will be in the order of several hundred milliseconds at best. This will only be true if he is not distracted – in reality, it may be several seconds before any gain reduction is imposed on the signal to bring it under control – not an ideal method of protecting the system from damage.

The first solution is to introduce an electronic form of system gain control – an analogue dynamics processor. Immediately, using an analogue dynamics processor (in this case, a limiter), the situation is much better. Controlling the gain electronically, and not relying on a human sidechain feedback mechanism, it can react much more quickly.



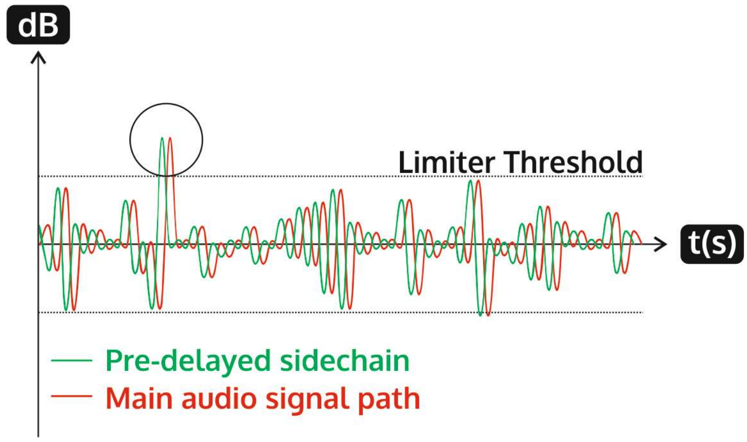
The red waveform represents the input to the dynamics module, with the dotted line showing the threshold for gain control to occur. Peaks above the threshold should trigger a response from the limiter, and so the dynamics processing should react to these as appropriate. (In this case reduce the gain).



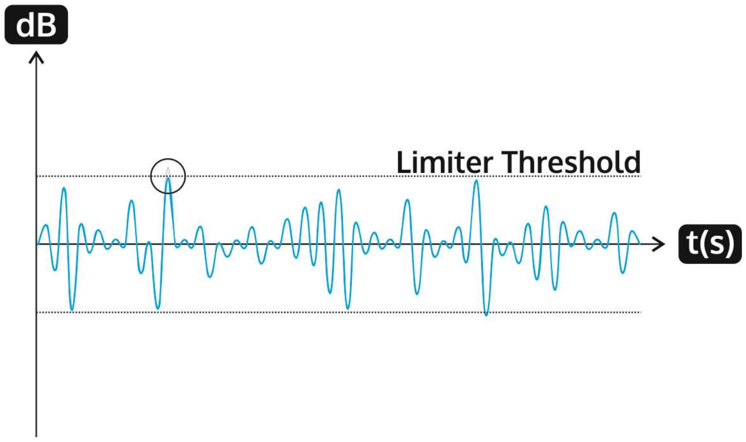
The blue waveform shows the output of the dynamics module. The circled peak demonstrates that the processor has partially "missed" the first peak above the threshold (as it is very fast and short), but has "caught up" shortly afterwards, keeping all other peaks under control. As it is unable to predict what is coming, this will always be a failing with analogue dynamics processing.

In the digital domain this problem can be efficiently addressed. The peak limiter pre-delays the sidechain signal, resulting in a "zero overshoot" limiter, which is able to catch all peaks and provide a reliable absolute maximum setting for the output of any channel.

Note that this is a *pre-delay* – which means the processor takes advantage of the fact that there is signal processing before the limiters in the signal path and can use a monitor point for the sidechain control signal some samples earlier (in our case, 16 samples earlier, or approximately 17uS) – there is no additional audio delay introduced to implement the clip or peak limiter.



The pre-delayed sidechain is shown in green, with the main signal in red. As the main signal now exists slightly later in time compared to the sidechain, the output from the limiter does not suffer from the overshoot problem. This is because it can begin to react ahead of the main signal's peak and so control it correctly. This is shown in the limiter output below - no overshoot even on the first circled excursion above the threshold.



Remember that this delay is only in the order of tens of microseconds,and is a pre-delay – the sidechain is moved back in time in relation to the main signal. Inserting a delay into the main signal path of an analogue dynamics processor will achieve similar results, but with the penalty of delaying the main signal by the amount of look ahead delay introduced.

The clip limiter follows the RMS limiter, has only two parameters to adjust – the release time and the threshold. Note that the threshold is set to be a minimum of 2dB above the threshold of the program limiter – setting the threshold to "10dB above" means that no more than 10dB of overshoot above the threshold of the program limiter will ever be allowed.

The release time can also be automatically set if the RMS limiter has automatic time constants enabled and so are set by the high pass filter frequency for that channel.