WEISS DS1 DIGITAL DYNAMICS PROCESSOR

Gateway Mastering Engineer SCOTT McCONVILLE reviews the latest in the line of Gambit processors from Weiss Engineering.

Since 1984, Weiss Engineering Ltd has been in the business of producing digital audio processors for recording and mastering studios. The Weiss BW-102 series is a staple in mastering studios the world over. Based on a modular concept, components can be purchased separately and installed in a chassis, in whatever configuration is necessary. Modules available include digital input and output, equalisers, level controllers, compressors, de-essers, redithering, and other digital audio necessities. The BW-102 De-esser module is considered one of the finest sounding de-essers available. In 1996, Weiss began producing the Gambit series of standalone, rack-mountable, digital audio processors. I took one of their latest creations in the line, the DS1, to Gateway Mastering Studios in Portland, Maine, and put it to the test.

Overview

The DS1 model number seems to indicate that this dynamics processor is optimised for de-essing. That may be true, but a wide range of control gives the DS1 the versatility for almost any dynamics application. The split filter can be widely adjusted for band-selective compression/limiting. Turning off the filter enables the compressor/limiter mode, and allows the DS1 to operate over the full audio range. Four-times upsampling is provided for high-speed transient detection, and double-sampling signal processing is implemented for the entire compressor section. Snapshot memory, with A/B switching,' allows for quick comparison of set-ups. Remote control is via MIDI, RS-232 and RS-422.

Features & Operation

The DS1 is a digital-only processor. No provision is made for on-board converters. The model evaluated supported 44.1kHz and 48kHz sample rates with a bit depth to 24 bits. Future versions will be provided with up to 96kHz capability, making full use of the internal double sampling capabilities of the unit. This could make the DS1 a valuable tool in the quest for high-bit-rate audio, destined for DVD (see sidebar). The Weiss DS1 shares many design characteristics with its Gambit series predecessor, the Weiss EQ1. The DS1, in a 2U high box, has a smartly laid out front panel with an uncluttered and efficient design. Weiss' push buttons and rotary encoders provide noticeably smooth and positive input. These controls, used on previous Gambit series boxes, have always provided problem-free operation. Menus, metering, and status display are provided by a liquid crystal display with a 'cold cathode fluorescence' backlight. That's a technical term for 'no flashlight necessary'! This type of display has much better off-axis viewing than other, more common, liquid crystal displays. Five push buttons to the right of the display allow the user to navigate through menus and select various options corresponding to the display. A single gain/data knob provides data entry for many of the menu choices. Eleven knobs are given dedicated functions; attack, release delay, release fast, release average, release slow, soft knee, threshold, ratio, gain makeup, frequency, and bandwidth. These knobs are touch-sensitive. A momentary touch will switch the display to indicate the appropriate settings for the knob. No need to move a control to get its attention. The display will switch back to its previous indication after a few seconds. At times, while adjusting and listening, this was a hindrance in that the display switched back before we had a chance to make up our minds.

Parameter Settings

A typical display, at first glance, appears slightly confusing. Upon further inspection, it is just displaying as much useful information as possible in a 3.5cm x 12.5cm display area. Parameter settings are shown in several ways. For example, an 'attack' setting will be shown numerically in milliseconds, and graphically as a transfer function (the typical level-in/level-out chart). It will also be shown on a bar chart for relative comparison to other parameters. Compressor stage input level and output levei is indicated by peak meters and is cleverly tucked into the display just above and to the right of the transfer function chart. If crossover filters are utilised, the frequency and bandwidth characteristics are indicated on a small graph across the top of the display. The all-important gain reduction meter is given prominent location to the left of the display. Meter response is surprisingly fast and responsive. Overall, the graphical display conveys enough information to confirm that your settings are in the neighbourhood, and how hard the box is working. Below the display are the push buttons for managing workspaces, snapshots and presets. The bypass button provides bit-for-bit transparent bypass. A red button activates the extremely useful feature of monitor mode.

This mode routes the signal being processed (side chain audio) directly to the output. It conveniently allows you to focus in on just the sounds that require gain reduction. All you hear through the AES output is the affected sound.

De-esser

A dedicated de-esser preset button accesses four de-esser presets on the display; hi-hat soft, hi-hat hard, vocal soft, and vocal hard. A limiter preset button gives quick access to soft, mid, hard, and bass boost limiter settings. The presets are adequate and provide a good starting point for experimentation. With such a limited number of presets, it won't take long to stray into the envelope detection and transfer curve parameters and start tweaking. The DS1 has two workspaces for temporarily storing custom set-ups. An A/B button allows quick comparison of each workspace. A workspace can be transferred to any one of 128 snapshot memories. This snapshot set can be backed up to either one of two, non-volatile memory banks. Understanding how a digital dynamics processor like the DS1 operates on the inside will save you hours of fumbling for the right sound. The DS1 has an adjustable digital delay on the main chain of audio. This delay buys time for the side chain to preview the signal, and apply a gain transfer curve to the main chain without missing the transients. What this means is less overshoot and distortion of the type typical in analogue compressors. It can give you the level control without damaging the critical transients that define the sound. Getting it right requires the preview delay to work in conjunction with the envelope settings. The delay time the DS1 applies is controlled by two settings – overall delay and preview delay. Overall delay is useful for timing the output of the box to other devices, which is critical when channels take diverging signal paths or require differing amounts of digital processing. Overall delay is adjustable from 2ms to 228ms. Preview delay is the time the side chain has to look ahead. Preview delay is adjustable between 20us and 200ms. Preview delay is always some fraction of overall delay, it can never be greater. Envelope parameter times are wide ranging enough for almost anything you might throw at it. Attack times start at 20ms and increase to 800ms. For best results, the attack time and the preview delay time should be considered together. Weiss recommends setting the preview time to approximately three times the attack time. The delay should allow time for the side chain to catch leading edges. Theoretically, while using preview delay, the side chain has time to see all the transients and act without passing nasty over levels. If shorter delay times are used, a gain makeup control is available to counter any overs that might get through.

Release Times

Setting release times on the DS1 requires some explanation. Four controls are provided to adjust the release parameters. These are: release delay, release fast, release average, and release slow. Each can be adjusted from 20ms to 8s. The release delay determines how long the DS1 waits before applying the release time parameters of the envelope. This helps reduce pumping effects by reducing premature gain changes. The DS1 will dynamically choose the release time depending on the program material. The input signal is continuously monitored and measured for peak and RMS amplitudes. The ratio of these readings determines the response characteristic. The release average parameter is the time period over which the program material is averaged. Fast 'average settings' will cause the gain reduction to track quickly and use the 'release fast' setting. Slower 'average settings' will use the 'release slow' setting more towards the end of the envelope. To help sort this out, you'll need to watch the gain reduction meter. Release settings will have an effect on how fast the gain reduction meter responds. The transfer curve settings are soft-knee, threshold, ratio, and gain makeup. The threshold, the level that must be reached before gain reduction is applied, is variable between 0 and -60dB in 1dB steps. Ratio values between 1:1 and 1000:1 are available. Soft-knee settings give the compressor a more gradual change between ratios, similar to analogue processing. Soft-knee can be turned off or varied between 0.1 and 1.0 in 0.1 steps. The gain make-up ranges from -10dB to+27dB in 1dB increments. Filter centre and cut-off frequencies start at 277Hz and end at 17.7kHz. The DS1 utilises FIR filtering and is totally phase linear Bandwidth is indicated in octaves and ranges from 1/6 octave to six octaves. While in a band-selective mode such as de-essing, the DS1 offers absolute transparency. This feature, especially beneficial to mastering engineers, provides bit-for-bit transparency when no gain reduction is being applied. The output of the DS-1 will be a bit-for-bit clone of the input until the signal requires processing. This reduces the unwanted cumulative artefacts of the effects chain. When operating in full-range mode, the DS1 is no longer bit-transparent due to the up- and downsampling process. An overall gain adjustment is available and permits raising the gain 10dB or lowering to -90dB and infinity. We were surprised to discover that gain adjustment on a dynamics processor sounded better than on other digital level controllers and workstations! Dithering is second order noise shaping with an auto-blacking feature. Auto-blacking will turn off the dither while the input signal mutes, thereby maintaining digital zero.

Conclusions

Don’t expect to get optimum results without reading the manual. Don’t expect optimum results immediately after reading the manual! To reach the full potential of the DS1, expect to spend time spinning the knobs and listening. The DS1 is very musical-sounding. It’s remarkably clean-sounding. Weiss has given a lot of thought about what a dynamics processor can do, and it shows in the amount of control built into the DS1. The DS1 has the finesse required for demanding mastering applications. Weiss Engineering’s experience in working with the people that use their products has given them the insight to design forward-looking products. Future 96kHz support will allow greater flexibility for the ever-increasing demand of high resolution multi-channel sound. The 96k upgrade promises to give the DS1 more DSP power and the ability to link three units together for 5.1 processing. We’ll take three!

5.1 READY?

These days, multi-channel music is on the mind of many mixing and mastering engineers. Until now, we have been able to get by using standard processing and borrowed surround tools from the film industry. DTS releases on CD, and video releases on DVD, have created a need for high-quality digital processors that can handle the special demands of the new audio formats. Current DVD titles being produced with the DVD Video specification are not limited to 5.1 Dolby Digital. Two-channel 96kHz/24-bit stereo PCM is also supported, and several music-only titles are available. All of the DVD players on the market play 96/24 off the disk (however, most will decimate down to 48kHz before hitting the converters). The new DVD-Audio specification 1.0 promises many more formats – and challenges! DVD-Audio’s lossless compression offers consumers six channels of 96kHz, 24-bit audio. 192kHz 24-bit stereo is also supported. Mixers and mastering engineers are creating a demand for digital signal processors flexible enough to support these multiple sample rates. Boxes built for stereo need the ’smarts’ to work together across multiple channels. Not always as easy as it might seem. MIDI interfaces help to sort out the control problems but anomalies can still creep into the system. Digital processors take time to do their calculations. Adding a filter on the left and right channels may introduce a delay that the centre channel doesn’t see – and you’ve got timing and phasing errors. A processor with an adjustable delay time can compensate for the ’stereo’ boxes it doesn’t see. Just being synchronised won’t cure all the problems either. Some processors dynamically adjust their algorithms over time to optimise for the program material. Listen to two channels and it sounds great; listen with all six and subtle phasing errors may periodically swirl around you. Differences in mixing styles may influence the effectiveness of compression and de-essing with multiple stereo boxes. Shifts in front/back imaging may occur if processing channels are grouped in pairs, and instruments are panned between them. One box will not know what gain reduction the others are applying. High bit-rate audio poses some interface questions as well. Early equipment designs have seen single connector AES, dual connector AES, High Speed SDIF2 and optical connections. In the near future, the safe bet might be to have all of the above. Equipment manufacturers, having designed their box for stereo, may not yet be aware of how it works with 5.1 mixes!