CEDAR Audio Duo Auto Dehiss

Broad-band Noise-reduction Processor

By Hugh Robjohns Published May 2005

Photo: Mike Cameron

This slimline rack processor makes CEDAR's latest class-leading audio-restoration algorithms remarkably quick and easy to use.

For anyone involved in the specialised crafts of audio restoration and 'forensic audio', CEDAR Audio will be a familiar name. This Cambridge-based company have long been the standard bearer for the kind of highly specialised digital signal-processing tools required in these professions. In recent years, the advances in computer processor speeds and general DSP functionality have enabled many manufacturers to develop their own audio-restoration programs, and many are very cost effective, but to date no-one has quite managed to match the class-leading functionality and quality of the highly evolved CEDAR algorithms.

Most of CEDAR's restoration tools are available on a variety of software and hardware platforms. The company's flagship software platform is called the CEDAR Cambridge, a PC-based system which can be equipped with a wide range of the latest signal-processing 'modules' to suit a variety of restoration and forensic applications. Many of the same core modules are also available as plug-ins for DAW platforms including AMS Neve workstations, Merging Technologies Pyramix, Studio Audio & Video SADiE, and Sydec Soundscape. However, this review is of CEDAR's latest generation of hardware, the new Duo series, which effectively supersedes some of the previous Series X rackmount units.

Action Stations

The Series X Dehisser was called the DHX (reviewed in SOS July 2000). It boasted a 50 megaflops DSP engine, with all-digital I/O and three simple rotary controls to adjust the critical dehiss algorithm parameters (level, variance and attenuation). By comparison, the new Duo Auto Dehiss reflects the advancing technology of the last five years very clearly, both in terms of its looks and its specifications.

The first impression when unpacking the unit is that the steel case has a very similar design to that of its forebear, with three horizontal engraved lines running across the full width of the matt-black front panel and highlighted with a subtle gloss-black CEDAR logo. The case extends a modest 200mm behind the rack ears according to my ruler (although the handbook claims 240mm), but the new model is almost twice as heavy as the older DHX, tipping the scales at 4kg.

As on the DHX, the rear panel carries only a few connectors: a pair of XLRs for stereo AES digits in and out, a pair of phono sockets for S/PDIF in and out, and a trio of MIDI sockets (In, Out, and Thru) for remote control. One new addition is a USB socket (apparently for factory use only), and the IEC mains inlet feeds an internal universal mains power supply accepting voltages from 85V to 260V at 50Hz or 60Hz.

Whereas the DHX could only operate at sample rates of 44.1kHz or 48kHz, the Duo accepts any sample rate between 32kHz and 96kHz. The I/O supports 24-bit word lengths (although the internal resolution is 32-bit), and the available DSP power is an impressive eight times greater than on the DHX — a pair of Sharc DSPs providing 400 megaflops of signal-processing power.

The front panel also reflects half a decade of technological advances. For a start, the old-fashioned green power LED of the Series-X models has been upgraded to the new millennium's omnipresent blue indicator. In fact, there are two blue LEDs at the right-hand end of the front panel: one marked Power, and the other Standby. As on the DHX, there is no mains isolation switch — the unit is permanently powered as long as it is plugged in — but a front-panel rocker switch enables a power-saving standby mode. Another sign of the times is that, while the operating power consumption of the Duo is the same as that of the DHX at 15W, the standby mode curtails the power consumption to just 1W instead of the 10W of the older DHX.

Possibly a slightly less welcome change for the Duo is the replacement of the DHX's three simple physical controls with a blue (what else?) LCD menu screen, six soft keys, and a rotary encoder. The screen is very clear and the menu operation simple and intuitive, thanks in part to the bright/dim illumination of the backlit blue buttons to indicate which functions are available and which are not.

Photo: Mike Cameron

The rear panel carries a pair of XLRs for stereo AES digital in and out, a pair of phono sockets for S/PDIF in and out, and a trio of MIDI sockets (In, Out, and Thru) for remote control. The USB socket, apparently, is for factory use only.

Setting Up

After you switch the unit on, the LCD screen shows the CEDAR logo and the product serial number for a couple of seconds while the software boots up. After this, the main operating menu is displayed, with the three principal user parameters on the left-hand side, and mode and menu options on the right. The middle right soft key accesses the Menu mode, providing six new options labelled Setup, Audio I/O, Process Mode, Memory, Close, and Status — and I suspect the functions of most of these will be self-evident. The Setup menu allows the screen contrast and MIDI channel number to be changed, as well as displaying the software version and hardware serial numbers. There is also a facility here to clear the user memories and restore the factory defaults.

The Audio I/O menu enables the output word length to be selected (16, 18, 20, or 24 bits, with TPDF dithering) and the output gain to be adjusted over ±10dB. There is also a numerical signal-level display with a peak-hold function. Strangely, there is no means of selecting a specific input — the unit apparently selects whichever has a valid signal. So what happens if there is a valid signal on both? The handbook recommends against this to avoid an incorrect automatic selection being made! Both outputs are active at all times, and any status bits present on the active input are mirrored to the outputs. The Status menu indicates the input connection and clock lock status, along with the measured input sample rate and the current state of the two DSP chips (indicated as either OK or Error, the latter with numeric fault codes).

The Memory menu provides access to the 99 user memories, with the usual Store, Recall, Rename, and Delete functions, while the final Process Mode menu allows the unit to be configured for manual or automatic operation, and to process stereo signals in either L-R (left-right) or M-S (Middle & Sides) formats. This menu is, however, mostly redundant as these configuration modes can be changed directly from the main operating screen.

In the default automatic mode, the screen shows the settings of three adjustable parameters on the left, each accessed by pressing the adjacent soft key and then using the wheel encoder to change the value. I'll return to these user parameters in a moment. The right-hand side of the screen shows the processing mode at the top (L-R or M-S) with underlines to indicate which channel is being processed. Repeated pressing of the adjacent soft key cycles between left, right, and both (or Middle, Sides, and both), and holding the button down for three seconds changes the mode between L-R and M-S.

Photo: Mike Cameron

The middle button always provides access to the configuration menus, as already described, while the bottom button provides a bypass function to switch the processing in and out to check its subjective effect. Holding this button down for three seconds will change the operating mode from automatic to manual, and vice versa.

In automatic mode, the software itself identifies the hiss element of the signal and optimises the various program parameters accordingly. The screen displays three adjustable parameters, labelled Bias, LF Bias, and Atten. Normally, the last is the only control that requires setting, as this determines the amount by which the hiss is attenuated. This is a subjective decision and will vary with the material and the desired effect.

The Bias and LF Bias controls allow the user to offset the program's automatic hiss detection. Setting a positive Bias effectively instructs the program that there is more hiss present than it has detected, and the output will therefore be processed more heavily. A lower Bias setting does the reverse, which helps to retain more ambience in the signal, although with a greater risk of noise artefacts slipping through. The LF Bias control does exactly the same, but only influences the system for frequencies below 5kHz — low and mid-range frequencies. This allows the hiss reduction to be tailored to suit the spectral character of the medium. An analogue tape, for example, where the hiss is most audible across the higher frequency range, might benefit from less low and mid-range hiss reduction, and the LF Bias control could be reduced accordingly.

Adjusting either of the Bias controls normally requires the Atten setting to be tweaked as well, and there is an iterative process of optimising the bias settings and fine-tuning the attenuation to get the best results. However, the effect of each control is quite audible with a little experience, and in practice setting things up is reasonably quick.

Photo: Mike Cameron

CEDAR's previous generation of 1U rackmount audio restoration tools were the Series X, reviewed back in SOS July 2000. The march of progress has been swift, though, and the Duo processors now provide eight times the processing horsepower of the Series X units, as well as a more flexible LCD-based control interface.

In manual mode, the menu screen looks much the same. The three right-hand buttons perform the same functions as before: channel selection, menu access, and bypass (with the same 'long press' options working as before). The three left-hand parameters initially look the same too. The Atten and LF Bias controls are exactly as before. However, in place of the Bias control the manual mode provides a Level control. This is used to instruct the software of the absolute level of hiss within the signal, and operates across the entire audio spectrum. In effect, this is the parameter which is set automatically in automatic mode.

Clearly, this parameter is critical to the effective operation of the dehiss process, although finding the optimum position is actually quite intuitive. If set too high, low-level ambience is treated as noise and removed, resulting in an unnatural and muffled sound quality. If set too low, the device will not remove noise effectively at all, and you may start getting noise artefacts. Again, an interactive process is required to fine-tune the Level, LF Bias, and Atten parameters, but finding the optimal settings doesn't take long.

The facility to process the signal in M-S mode is very useful. In many cases the strong central component of stereo material may mask hiss quite effectively, while at the edges of the image the hiss may be more exposed and obvious. Processing in M-S mode allows the hiss reduction to be tailored accordingly across the stereo image. Likewise, in L-R mode if there is more noticeable hiss on one channel than the other, or if the two channels exhibit different 'colours' of noise, the program can be optimised accordingly.

Performance

The Duo Auto Dehiss is certainly an impressive tool, and the algorithmic improvements over the previous generation are obvious, both when setting it up and when listening to the processed output. This new version seems more assured of what to class as noise within a signal, and the distinct 'twittering' and 'glugging' artefacts incurred while adjusting the Level parameter of the Series X machine have been replaced with a more intuitive noise-pumping effect.

No dehiss program will ever be able to remove every trace of noise perfectly while leaving low-level ambience and reverb tails completely intact, but the Duo Auto Dehiss comes very close to that ideal in a fuss-free way. I used the machine to clean up a selection of classic jazz recordings on quarter-inch tape, some dating back nearly forty years. The results were very good indeed, and optimising the settings was always intuitive — although it certainly pays to fine-tune the Bias, LF Bias, and Level controls in a reiterative loop a couple of times, because adjusting one seems to have knock-on effects for the others. The more care and attention paid during setting up, the better the end results, as you might expect.

While a lot of the material was mono, I found the M-S mode very useful with some of the stereo tapes, and the ability to bypass the hiss removal processing made it easy to assess the effect on reverb and ambience, as well as making it easier to recognise any low-level artefacts. I suspect longer familiarity with the product would result in very effective 'ear-training' which would allow the optimal settings to be found even more quickly.

Like all CEDAR products, the new Duo Auto Dehiss is relatively costly — this kind of R&D is expensive to fund — but it provides a real-time solution to the problem of unwanted hiss. The automatic mode seems to optimise the settings very well, and there are no audible artefacts from the processing if you use it sympathetically. Overall, a very worthwhile improvement over its predecessor.

Pros

- Fast, accurate automatic configuration, with flexible fine-tuning facilities
- Options to process channels separately or in Middle & Sides format.
- Phenomenally effective noise-reduction algorithm.
- No audible side effects if used sympathetically.

Cons

- This degree of sophistication doesn't come cheap.

Summary

The CEDAR Audio Duo Auto Dehiss represents another step forward in the technology of hiss-removal tools. Both the hardware and the algorithm have benefited from updating, enabling this new unit to offer even more impressive performance controlled through a remarkably simple but flexible user interface.

Information

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