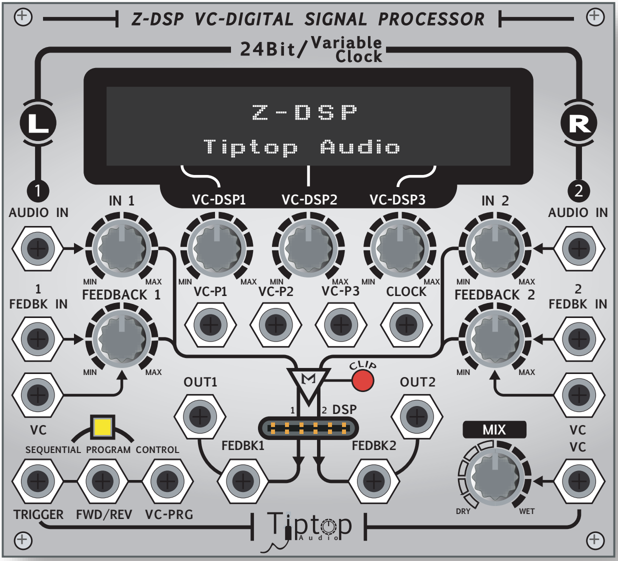
Tiptop™ Audio | # Z-DSP



User Manual

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Z-DSP VC-Digital Signal Processor

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MADE IN THE USA

Contents.

Introduction.

Welcome to the world of digital signal processing! The Z-DSP is a modular synthesizer component that can process and generate audio using a dedicated micro-computer, a digital signal processing processor – a DSP!

Like the processor in your desktop computer, the Z-DSP runs programs in memory. It's these programs that create the delays, filters, oscillators and more that the Z-DSP can produce. The possibilities are virtually endless, limited only by the imagination of those who write the applications for it. Tiptop Audio, together with some of the best known programmers in the music industry, are working to bring programs to the Z-DSP platform.

The Z-DSP uses an open-source coding environment, and using a programmer (available from Tiptop/Spin Semiconductor) anyone can create, share or sell their own applications for the Z-DSP. We hope that this unique feature will motivate more designers and users to dive in to the amazing world of digital signal processing and enrich the available library of applications for the Z-DSP platform. Please contact us for more information regarding obtaining a programmer.

Getting Started.

The Z-DSP itself contains no programs, it loads programs from a cartridge. If the module is powered up without a cartridge inserted it will show "insert cartridge" on the display

The module is sold with the Dragonfly Delay cartridge, which contains 8 delay programs. Pull it out of the bag now and insert it slowly to the card slot on the front of the Z-DSP, making sure that the Dragonfly print is facing upward. Note: Inserting the card upside down will not cause any damage, but the card will not work.

After a moment, the Z-DSP will load the program found in memory slot 1 on the cartridge. Some cartridges will display a message when first inserted. For example, the Dragonfly Delay will first show the name and author of the algorithms, then show a reminder that audio needs to be connected to both inputs to achieve a stereo effect. Note: A Tiptop Stack Cable is ideal for bridging inputs!

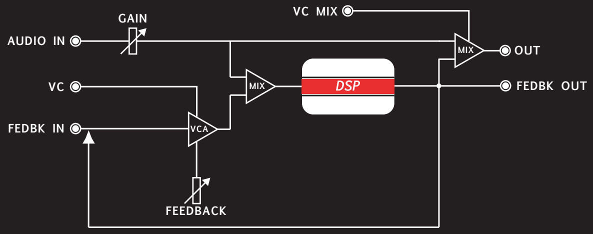
The cards can be inserted and removed at any time, even during audio processing. Pulling the card out at this point will keep the current program loaded and the display will again show "insert cartridge". Now that we have a program loaded into the processor, let's have a look at the Z-DSP signal flow.

Signal Flow.

The Z-DSP contains two distinct audio channels, labeled Left (also “1”) and Right (also “2”). The terminology of “Left” and “Right” is most commonly used for stereo effects like Delay and Reverb, while “Channel 1” and “Channel 2” would be used in applications that deal with more diverse names such as Carrier and Signal in a ring modulator.

The Z-DSP is truly stereo, no summers allowed! Each channel is built from a distinct audio input , feedback input, processing block, audio output and feedback output.

SINGLE CHANNEL IS SHOWN



Analog Feedback.

Feedback is the process of taking an output and applying it ("feeding" it) back into an input. This technique is widely used in audio for a variety of applications and is an especially strong tool in DSP allowing samples to be re-processed.

The Z-DSP offers an open-loop-feedback architecture which means that the user has the freedom to insert other processing devices in the feedback loop.
For example, analog filters, frequency shifters, other DSP processors, etc.

The Feedback Input contains a VCA that allows control of the gain of the feedback loop. Given that this is a VCA, you can control the gain from any voltage source. The VCA is very responsive to control input and can be swept up into the audio range for even more wild feedback effects.

The feedback loop on the Z-DSP is hardwired internally so with nothing plugged into the Feedback Input jack, it is fed from the 100% wet Feedback Output. Turning the Feedback Input knob clockwise will introduce more signal back into the input of the channel. Inserting a plug into the Feedback Input jack will break the loop.

The feedback section has a good amount of gain in it, and will easily cause the module to self-oscillate. This can result in some high frequency 'screeching' which can harm your monitors. So take it easy on that gain knob if you're looking for smoother sounds.

Digital Feedback.

Some algorithms process feedback internally, in the digital domain. The texture of digital feedback is much different and brings a very different flavor than analog feedback. Combining analog and digital feedback will bring even more depth to a sound.

You'll know that a program is using digital feedback from FDBK showing on the display as a parameter.

Reading the Panel.

The Z-DSP front panel contains graphics and typography to help you understand the signal flow and to indicate the functions of the knobs and jacks. Some shortcuts used are:

FEDBK or FDBK	Feedback
VCP1	Voltage control digital Parameter 1
VC-PRG	Voltage Controlled Program
FWD/REV	Forward/ Reverse
I/O	In / Out

From top-to-bottom/left-to-right the panel contains:

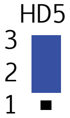
- ⊙ LCD display
- ⊙ Audio inputs jacks and knobs
- ⊙ DSP parameter control knobs
- ⊙ Feedback input jacks and knobs
- ⊙ DSP parameter control CV inputs
- ⊙ Clock/Sample rate jack
- ⊙ An audio clipping led indicator
- ⊙ Audio and Feedback Output jacks
- ⊙ DSP cartridge socket
- ⊙ A Wet/Dry knob and CV jack
- ⊙ A program select switch along with 3 jacks for sequential program switching using either VC or trigger/gate signals

In total there are 8 pots and 18 jacks.

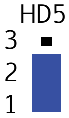
A built-in Guitar Preamp.

The Z-DSP contains a simple monophonic Guitar preamp that can be enabled by 2 jumpers at the back of the unit.

The first jumper allows a choice between synth (line) level and guitar level.

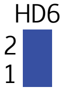


2&3 = synth level



1&2 = guitar level

The second jumper enables the preamp for both inputs, or just the Left input.



enable preamp for both inputs

In any case, a guitar should be connected to the Left channel input only and let the second jumper control the routing to the Right input.

Looking into Voltage Control.

The Z-DSP contain 2 types of Voltage Control (VC), one is the regular analog control such as the Feedback gain and VC over the Wet/Dry mix. These can be swept at any speed and well into the audio range. The circuit is designed such that with the knob at the center of its rotation, feeding a +/-2.5V signal will sweep the parameter from 0-100% for the Feedback gain, or 100% Dry to 100% Wet for the Wet/Dry mix.

The other type of VC is digital. This control input takes the analog signal and converts it into digital data. Voltage Control of the three DSP parameters is an example of this method.

The digital VC signals are filtered and smoothed to ensure that vibration, noise or supply variations do not cause the value to flutter between adjacent values. While this results in a smooth, noise free parameter control, there is a response delay of ~100ms. The response of these 3 inputs is very much like a Vactrol input in an analog module.

The 3 VC-DSP knobs allow for manual sweep of the digital parameters. The knobs act as an offset for the VC-P voltages, much like how the Frequency knob of an analog filter offsets a CV input. With the knob at its center position, a +/-2.5V signal will sweep the associated parameter will from MIN to MAX

The VC-PRG gives the user the option to switch through programs using different voltage levels. This

Looking into Voltage Control. - Cont

input accepts positive voltages from 0-5V though higher values are fine and will not damage the module. For some sonic chaos, try pulsing this input from an analog sequencer such as the Z8000 Matrix Sequencer/Programmer. More on this in the Program Switching section.

Overall the Z-DSP inputs are well protected against excessive voltages on the inputs. We do recommend however to stay within a reasonable range especially in case of using the Z-DSP with modular systems of different format and power supply as the Eurorack standard.

Understanding Clocking.

Probably one of the most powerful features of the Z-DSP is the CLOCK input.

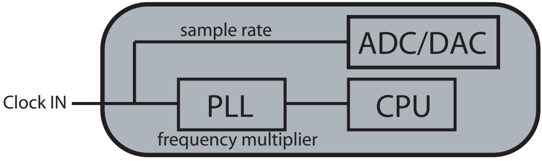
Your computer works by having a processor (CPU) execute lines of program code step by step. The processor runs at a speed that is controlled by a very fast clock. Your PC is running so fast that working with the machine is continuous and smooth.

But what if you could control the speed of this clock, making it slower and slower until it almost stops? How will your computer behave then, how will your software work, will it sound or look the same?? Will it crash!!!??

While we wouldn't want to try this on your PC (there are complications with devices like hard drives), we most certainly want to try it with our dedicated audio processor!

We have broken the rules and have allowed you to clock our DSP computer any way you like. The result is a fascinating feature that can turn standard effects into crazy things that you would never have expected.

Let's get started and understand what's happening here.



DSP CHIP

Understanding Clocking. - Continued

As we've discussed, the Z-DSP has a built in DSP processor. Along with the processor is a pair of 24bit analog to digital convertors ("ADC") on the inputs and a pair of 24bit digital to analog convertors ("DAC") on the outputs. The ADC samples the analog audio signal into digital data, while the DAC takes the digital data and converts it back to analog form. The programs that run in the Z-DSP work on this digital data, just like you would run a program on your computer to crunch data for your taxes.

The clock on the Z-DSP is what controls the speed of the DSP. In normal conditions, this clock runs at 32khz (the "sampling rate") which is fast enough to allow the ADC/DAC pair to provide 15Khz of bandwidth. The DSP uses this clock as well, but multiplies it to create processing speeds fast enough to run programs and keep up with the flow of data from the ADC.

This is a standard DSP clocking mechanism with a clock at a fixed frequency, and as long as nothing is plugged into the Z-DSP CLOCK input, this is what the Z-DSP will provide. That's all about to change...

By using the CLOCK input of the Z-DSP, we can change the sampling rate of the ADC and the associated speed that the DSP is processing data. That allows us to slow down the ADC, or if we use a VCO to provide the clock, we can vary the processing speed across time... There is a lot of sonic exploration to be done here!

Understanding Clocking. - Continued

To start, let's use the square wave output from a Z3000 VCO. First set the PWM knob to its center position, then set the Frequency and Fine knobs to maximum. At this point the Z3000 is set well above the 20Khz range, which will be our new sampling rate.

Now while the Z-DSP is processing a fairly bright sound, plug the Z3000 square wave into the Z-DSP CLOCK input. Congratulations, you just took over the system clock and replaced it with the clock from your Z3000!

You probably heard a sudden drop in high frequency component of your sound. That comes from the new clock being slower (~20khz) than the internal clock (~32khz), thus the ADC is sampling the incoming audio less frequently, thus reducing the bandwidth of the Z-DSP.

Now slowly start reducing the frequency of the Z3000 and listen to what happens. At some point the sample rate gets so low, and the program execution speed gets so slow that the result is glorious digital madness!

Ready for more? Connect an envelope generator or LFO to the FM input of the Z3000 and sweep the frequency up and down.... Get the idea?

Try modulating the Z3000 various ways in sync and out of sync from the VCP, and you will cause ordinary digital effects to perform in an extraordinary unpredicted manner. It is all about dynamic clocking as oppose to fixed rate clocking.

Understanding Clocking. - Continued

Anything is game for the CLOCK input... Modulate the pulse width of your new clock; set it to a narrow pulse so that the detector in the Z-DSP is “hanging on the edge”; try mixing the output of multiple VCOs to create a random clock. As well as going slower, there is also a whole new set of effects that stem from going faster. Feel free to go as wild as you wish with this, you will not harm the Z-DSP.

Note: It’s possible that excessive manipulation of the clock might cause one or both channels of the DSP processor to crash. If that happens you can reset the processor by switching through the programs until you get back to your original program.

Program Switching.

The Z-DSP cartridges can contain up to 8 programs, each program being a set of mathematical algorithms that manipulate digital data. As mentioned above, the ADC brings an analog signal into the digital domain by capturing it repeatedly thus creating a sample.

The DSP allows for various operations to be applied to a sample. It can be multiplied by some constant number (providing gain or loss), added to another sample (mixing), stored in memory and read out at a later time (delay), and many other functions. By using combinations of these operations, we can create effects, filters (such as tone controls), compressors, limiters, and other audio processes.

The DSP will execute the same set of algorithms on each incoming sample, producing one sample out for every sample in. The algorithm is a list of mathematical operations to produce the desired result, and one or more algorithms constitute a program. These programs are downloaded to the processor where the processor will continuously execute the algorithms on the sample stream.

The Z-DSP allows you to load programs from a cartridge by either manually pressing the yellow button or by feeding it a pulse or voltage for automated control. The Z-DSP has a built in sequential switch that allows the user to switch programs forward (1.2.3...7.8) or in reverse (8.7...3.2.1) etc. A trigger or gate signal sent to the TRIGGER input will switch to the next higher program (wrapping from 8 to 1). If a gate signal is applied to the FWD/REV

Program Switching. - Continued

input the direction will be reversed.

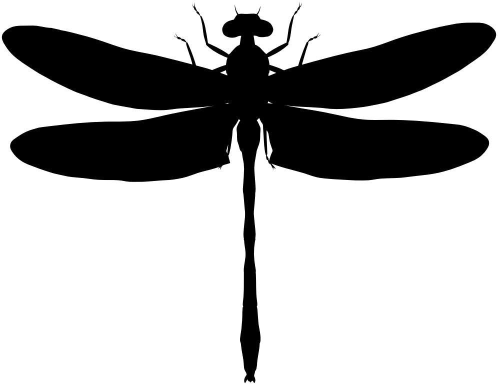
To control program switching from an LFO or envelope generator, use the VC-PRG input. A 0-5V voltage swing on this input will switch the program under the same terms as the TRIGGER input.

Note that switching time will vary from program to program. For example, switching to a delay effect takes longer than switching a filter effect. The delay effect needs to have time to fill the data buffer before passing the sound, and in some cases, this can take more than a second to complete. Filters however take a very short time to load and start working, since they do not need to buffer any data.

Another common thing with switching is switching noise (click). Switching noise is very much dependent on the effect used and on the audio that is being processed. The switching noise is more noticeable for example on sine waves than with pulse waves because of the lack of harmonic content.

Switching is a lot of fun and can add a rhythmic dimension, for example switching between filters using the Bat Filter card.

That should be enough to get you started... There's a lot to explore in the Z-DSP, don't be afraid to get funky with it!



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