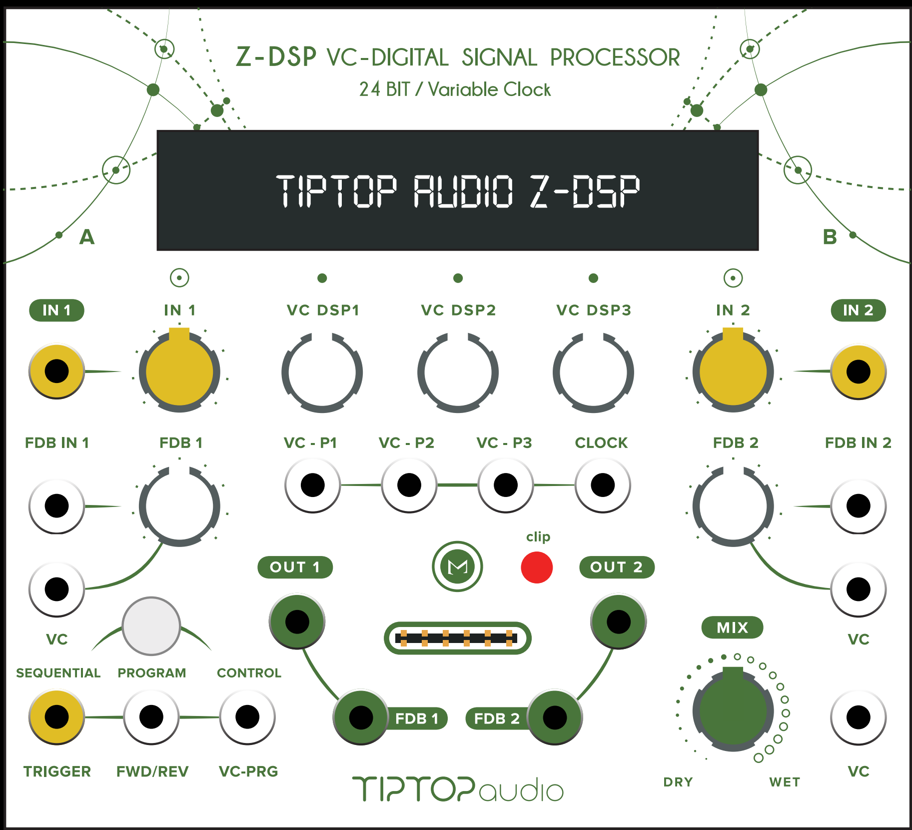
Z-DSP VC-DIGITAL SIGNAL PROCESSOR

24 BIT / Variable Clock



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Introduction

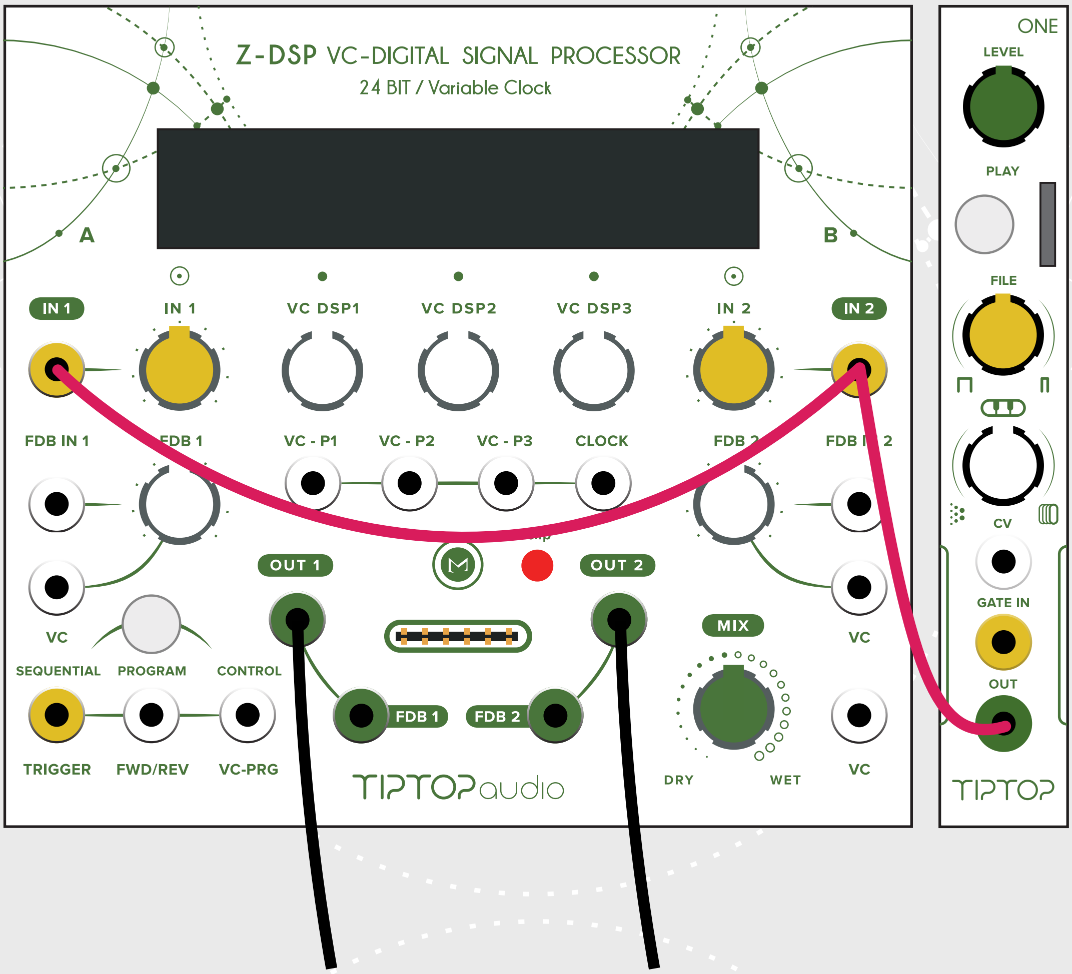
Welcome to the world of digital signal processing! The Z-DSP is a modular synthesizer component that can process and generate audio using a dedicated micro-computer, a digital signal processor – a DSP!

Like the processor in your desktop computer, the Z-DSP runs programs in memory. It’s these programs that create the sound effects 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 continually working to bring new programs to the Z-DSP platform. With over 10 years in the making, this library that is available on cards has grown very well and by now covers everything from classic reverbs and delays to experimental granular and physical modeling algorithms and a lot in between.

The Z-DSP uses an open-source coding environment, and using a programmer called NumberZ anyone can create, share, or sell their own effects for the Z-DSP. Checkout the Tiptop website for more information on NumberZ: http://tiptopaudio.com/zdsp-ns/

Unlike your desktop computer plugins the Z-DSP is based on a DSP processor chip surrounded by analog circuitry that gives it its warmth, and unlike your desktop computer the DSP chip can be CV controlled and its clock can be hacked, a feature that alter the effects in a unique and unpredictable way. Expect the unexpected. So let's get started.

Getting Started



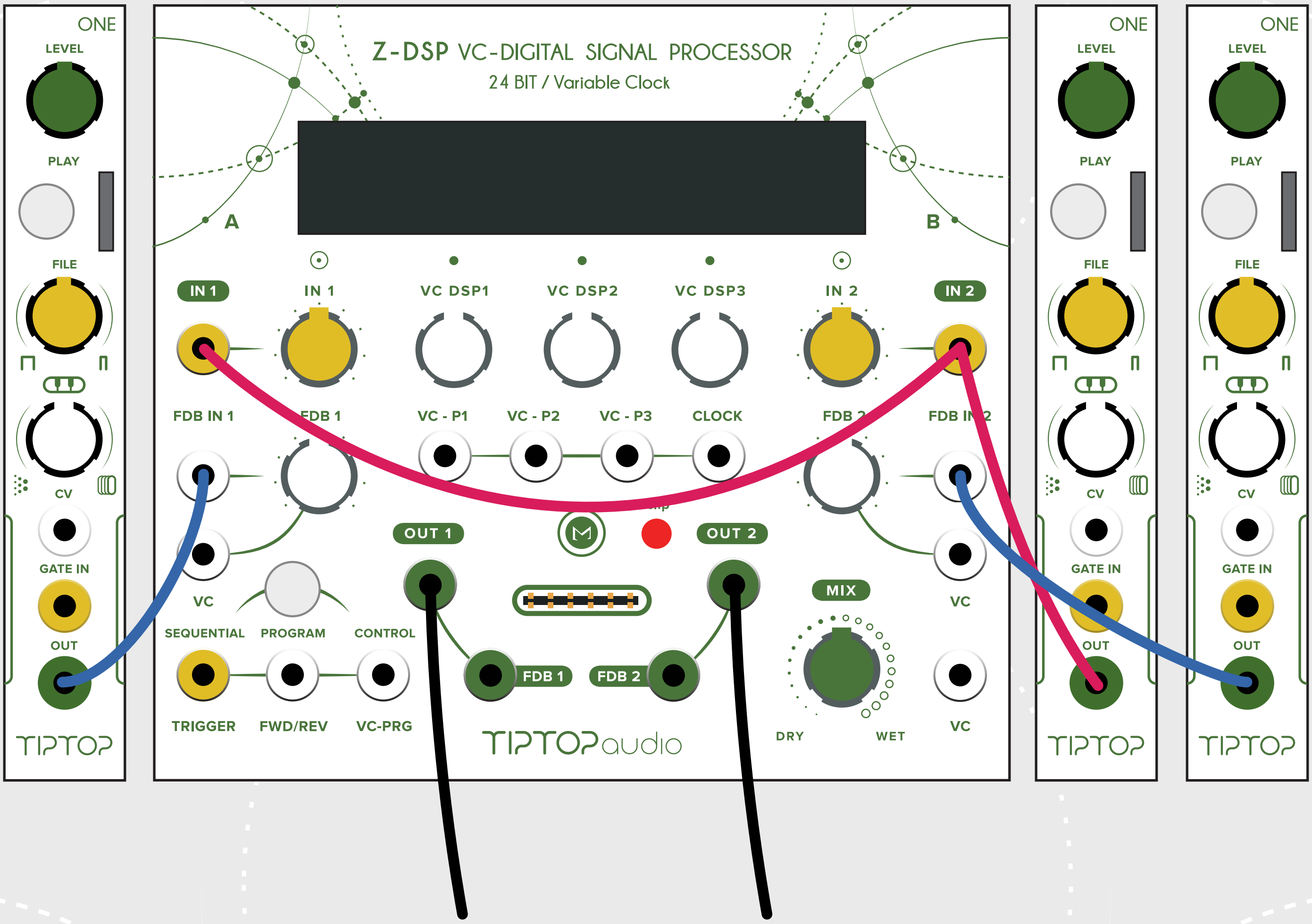
Using a Stackcable plug your sound source into both IN1 and IN2, set LEVEL knob 1&2 to half way, plug OUT1 and OUT2 to a stereo input on your external mixer or computer audio interface. The Z-DSP effects are mostly stereo so it's important to use both outputs.

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. On power up the CLIP L.E.D. might be on, that is normal and will go off once programs are getting loaded.

The module is sold with the Dragonfly Delay MKII cartridge and the Halls Of Valhalla reverb card from Valhalla DSP, each contains 8 algorithm programs. Pull the Dragonfly out of the bag 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 program one from 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 and both outputs to achieve a true stereo effect.

The cards can be inserted and removed at any time, even during audio processing and even when the module is heavily modulated. Pulling the card out at this point will keep the current program loaded and the display will again show "insert cartridge". With the card in, set the MIX knob halfway, set VC DSP 3 knob to max, set VC DSP2 halfway, set VC DSP 1 half way. Make sure both FDB1 and FDB2 are off. At this point you should be hearing a mono delay effect. Play with the VC DSP knobs to see what they do, slowly open FDB1 and FDB2 knobs to start add analog feedback, watch out as too much feedback and the effect will self oscillate, that can be loud. Change programs with the illuminated switch get a feel of the other effects in this card. If you have another sound source plug it onto FDB IN1 or FDB IN2 or both, see how these inputs can also be used to add additional sound source into the DSP wet signal.



Now that we played a bit, let’s have a closer 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 function such as Carrier and Signal in a Bit Rot logic 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.

Analog Feedback

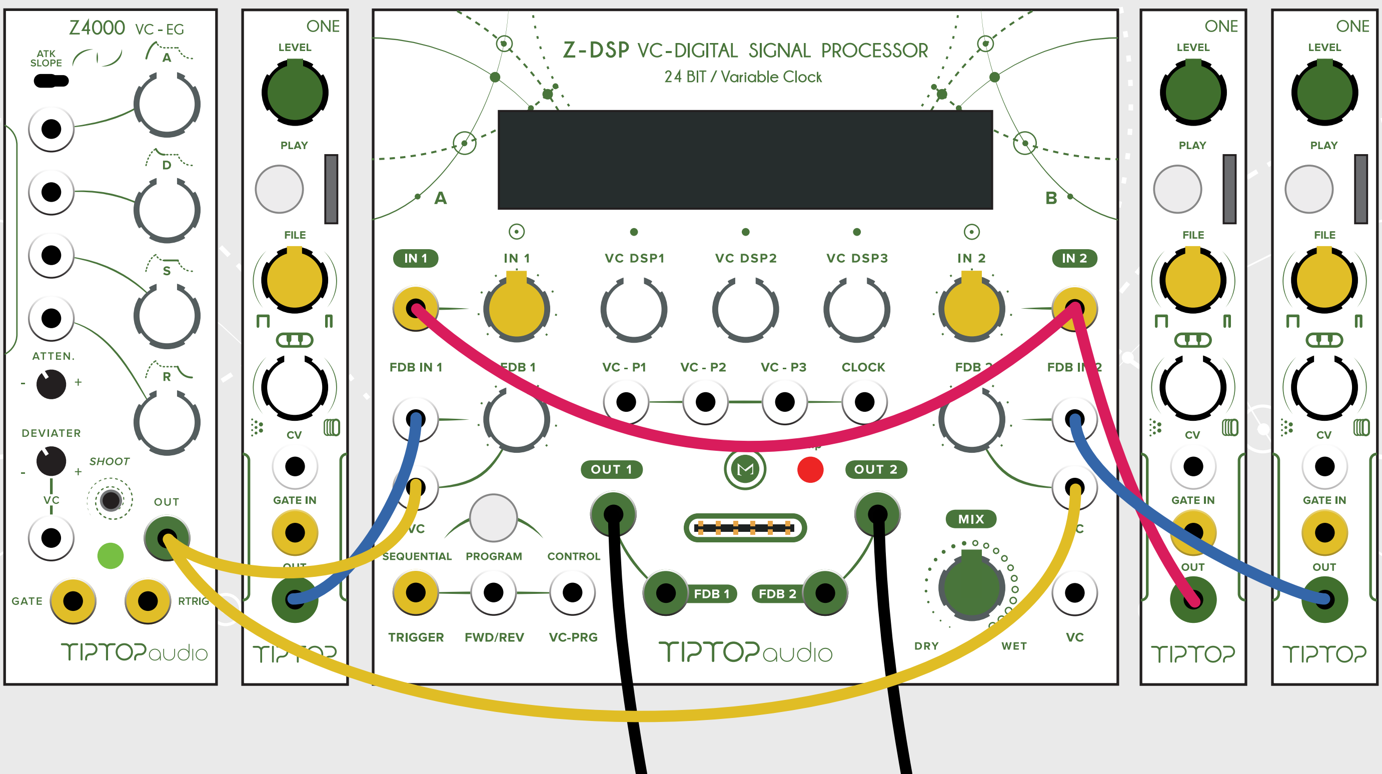
Feedback is the process of taking an output and mixing it ("feeding" it) back with the input. This technique is widely used in audio for a variety of applications and is an especially strong tool in DSP allowing samples to be re-processed again and again and again and..... Ok you get the point. On some effects it will make the 'tail' longer and longer. On others it will create a tone from self oscillation.

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 and other DSP processors etc. There are two Feedback output jacks labeled FDB1 and FDB2 (green) and two Feedback input jacks labeled FDB IN1 and FDB IN2 (white). These jacks are internally connected FDB1 to FDB IN1 and FDB2 to FDB IN2. If you plug a cable into the FDB IN, it breaks these internal connections and lets you use these inputs and outputs separately either to add inputs to to the Z-DSP or to route the Feedback externally through other modules. To place a module in the feedback loop plug one of the output FDB1 green jacks into that module input, plug the module output to FDB IN on the Z-DSP. Now you have an external module connected in the feedback path of the Z-DSP.

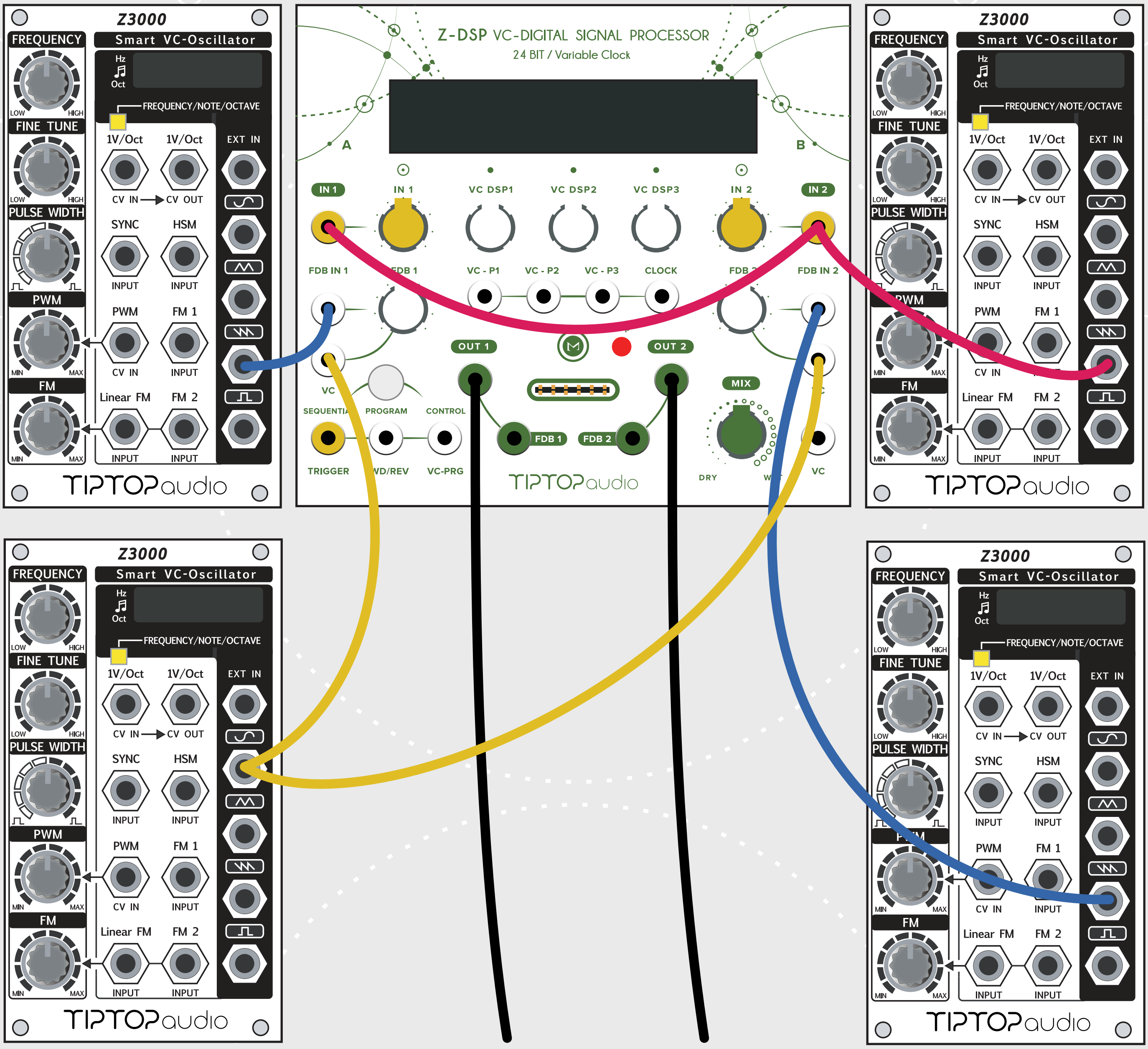
The Feedback Input contains a VCA that allows control of the gain of the feedback loop. Given that this is a VCA, you can control the gain from any voltage source. The VCA is very responsive to control input but can also be swept up into the audio range giving you a pair of analog modulators that can be used to implement Amplitude Modulation (AM) synthesis technique right at the entry into the DSP processor.

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 as explained earlier. 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 and turn this jack into another audio input for you to use.

Multiple inputs with VCA control



Built in VCAs Amplitude Modulation



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. Note that the FDB knobs are logarithmic so gain gradually adds up with the turn on the knob, it might be very quiet at the first few steps of the dial, this gives you smoother control of the input gain.

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 when Feedback or FDBK is shown on the display as a parameter. Some of the cards work great when using both analog and digital feedback enabled, give it a try.

Reading the Front 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:

FDB	Feedback outputs
FDB IN	Feedback Inputs
VC DSP1	Voltage Control Digital Parameter 1
VC-PRG	Voltage Controlled Program Switching
FWD/REV	Forward/ Reverse Direction of Program Switching
VC	Voltage Control Input

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 VC inputs
- DSP Firmware Clock/Sample rate jack
- An audio clipping led indicator
- Audio and Feedback Output jacks
- DSP cartridge socket
- A Wet/Dry knob and VC 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 knobs and 18 jacks.

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 are of that type.

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 Trigger input gives the user the option to switch through programs using a pulse.

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 a sequencer such as the Circadian Rhythms or Trigger Riot Matrix or just plug the pulse out of a LFO. More on this in the Program Switching section.

Understanding Clocking

For clarity lets first say that the Clock input on the Z-DSP is not for syncing delay effect to a BPM. This input takes in pulses at audio rate and up to ‘bend’ the effects in some unexpected ways. The ideal module to drive this input is the Z3000 square wave. The reason for that is because the Z3000 is a VCO that can go up to 31khz, which is higher than any other VCO in eurorack. Using other VCOs with much lower frequency range might not work well as these VCOs cannot clock the DSP chip at the speed where things start to get interesting.

Let's dive into the clock function to better understand what it does.

As we’ve discussed, the Z-DSP has a built in DSP processor. Along with the processor is a pair of 24-bit analog to digital converters (“ADC”) on the inputs and a pair of 24-bit digital to analog converters (“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 stream of digital data, very much like any audio effect processor in your computer.

The clock on the Z-DSP is what controls the speed of the DSP and ADC/DAC. In normal conditions, by the book, this clock runs at 32 khz (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! This technique is ‘illegal’ in engineering terms and is unique to the Z-DSP, yet it so powerful that it came to be known as one of the best things on the Z-DSP.

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 at about 31Khz, which will be our new clocking 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 your Z3000 as your new clock source!

You probably heard a sudden drop in high frequency component of your sound. That comes from the new clock being slower (~31khz) 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, something computers just can't do.

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 with changes on the the VC DSP inputs, and you will cause ordinary digital effects to perform in an extraordinary unpredicted manner. It is all about dynamic clocking as opposed to fixed rate clocking, a Tiptop Audio original.

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 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, many of them are group together to form further complex audio effects.

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 illuminated 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. 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 to 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 or bit crushers 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 binary ring modulators using the Bit-Rot 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!