The MXR model 113 digital delay – The creators perspective. V1.0
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Background

I designed the digital delay while working at MXR in the 70's. Keith Barr – my mentor, boss and friend allowed me to stretch my wings as a young engineer and do what couldn't be done, create a digital delay for under $1000 retail that would be useful to musicians and recording engineers.

Working in engineering, I had already developed the Phase 100, Noise Gate Line Driver, a rack with Power supply to hold the MXR Auto Phasor and Auto Flanger modules. One Saturday (Keith, Richard Neatrour and I would often work on Saturdays) I had begun to feed my urge to develop digital audio delay ideas. I had designed a 10 bit analog to digital converter (A-D) to experiment with. It used a technique called successive approximation. I used CMOS CD4013 flip flops (which we used in the MXR Blue Box) and some CMOS logic. The nice thing about this is that I would be able to listen to the converted audio because the A-D included a digital to analog converter (D-A). The circuit would sample the output of the D-A after a conversion which we could listen to. Would 10 bits be good enough?

On this Saturday Keith came in, asks what I was doing, I explain it and he was blown away. At this point, I was encouraged to continue working on this.

The world back then

The only echo units musicians could even hope to afford were magnetic tape based. These were mechanical, and unreliable. The tape loop had to be replaced frequently. The fidelity was poor. It would be great if we could find a way to create audio delays of any length with a purely electronic system.

Technology at the time was: Microprocessors were just being invented and of no use to us. Dynamic memory was just coming out and promised to be much cheaper, yet harder to use than static RAM. Small scale (SSI) logic was plentiful and cheap because of the computer industry. New medium scale parts were being introduced regularly. LSI was in it infancy. CMOS was coming on strong, but, it was slow. Computers were available as time-sharing services, one had to own or rent a terminal to access it. Scientific calculators were just out so we stopped using slide rules! No CAD. Pencil on paper drawings on drafting tables, a blue print machine, and hand taped PCBs on a light table.

Product goals

Over the course of the design the goals evolved. This is where the goals ended up:
- A musical effects product where time delay is the core of the effects.
- Capable of long echos to very short delays suitable for flanging.
- delay modulation to be able to do flanging and chorus.
- Suitable for use between a guitar and amplifier or as part of a chain of FX.
- Mixer to mix audio source with delayed audio.
- Regeneration (feedback) around the delay for echo repeats and enhanced flanging effects.
- footswitch bypass jack.

- Single control audio level matching, so that it could be used with a variety of applications.
- Built-in AC power supply.
- 2 unit rack mount due to weight and size.
- Starting price below $1000 US retail.
- 1 to 4 memory modules each allowing 320ms of delay for a total of 1280ms max.
- Repeat hold – a footswitch allows looping whatever is in memory without decay.
- x100 LFO for frequency modulation effects.
- Phase Inverted mix, noticeable when doing flanging effects.

The design

To create a digital delay required:
- Analog to Digital converter and Digital to Analog converter.
- anti aliasing and reconstruction filters,
- a compressor expander (to increase dynamic range of the delay) and a limiter.
- memory to store the digitized audio.
- Audio input output circuitry which included mixing, regeneration, level matching.
- digital control circuitry including a variable frequency high speed clock
- modulation of the delay with a low frequency oscillator.
- an AC power supply

A-D, D-A
The A-D is a critical component. Parts that you could buy at the time were primarily for instrumentation, weren't fast enough but most importantly way too expensive. We realized that the coding wasn't critical as long as the decoding matched. We were not doing digital signal processing of the audio. We came up with a scheme to do 10 bits of A-D, 5 bits at a time.
An input sample and hold grabs each incoming sample and holds it as a stable voltage while the A-D takes place. A successive approximation A-D is done using a D-A, logic and a comparator that compares the D-A to the input held at the sample and hold. We go through a set and compare process for each bit. When a 5 bit A-D is complete, we look at the difference between the input signal and our best 5 bit approximation of it in the D-A. This is amplified by 32 (2^5), the output of this error signal is the sampled and we do the same A-D process on that. Each of these 5 bit conversions are stored in RAM. When the data is retrieved, the 5 primary bits go through the same D-A that was used for the input conversion. The second group of 5 bits runs through the D-A, reduced by a factor of 32 and recombined with the primary sample to create a 10 bit result.

This technique offered a number of advantages. Doing 5 bits at a time helped simplify the data paths. The D-A converter only had to be 5 bits accurate because errors were compensated for by the second A-D cycle. Having the same D-A for both the A-D and D-A processes meant that errors canceled out. Having only a 5 bit A-D was much lower cost than trying to make a 10 bit system. Splitting the 10 bit A-D in to 2 5 bit chunks was Keith Barr's idea. I figured out how to actually do it. This was key to achieving our cost goals.
The A-D had to be fast performing up to a 200 kHz sample rate. Trying to find an op-amp that would settle fast enough was a dead end. I had to design my own with discrete transistors.
{For op-amp historians, the LF356 was the fastest settling op-amp I could find. The LM318 was the "fast" op-amp of the day, but it never settles. My discrete amp worked great because it wasn't hindered by being a marketable op-amp as far as input characteristics and drive.}

Having a master clock that was variable over a 4 to 1 range made a lot of the design challenging especially in the area of the A-D and D-A. We didn't actually get 10 bits of performance. The factor 32 above actually ended up being something like 24. I think it gave us around 9+ bits of performance.

anti aliasing and reconstruction filters,
In order to achieve long delays, we would limit the bandwidth of the audio. There are many music applications where lower bandwidth is very acceptable. So the bandwidth was one of 4 ranges: 20, 10, 5, and 2.5 kHz.
In order to prevent aliasing, which is an artifact of sampling, sharp low pass filters were needed. I had a 6 pole Chebyshev aligned input filter and 4 pole reconstruction filter. These were made up of 2 pole Sallen and Key stages. CMOS switches selected resistor values for each bandwidth setting. These are very complex filters to design. It stretched my math abilities considerably. I wrote programs on a time-sharing computer in the language APL to help design the filters. My programs were able to optimize the component choices to various tolerances, usually 5% parts. It would then run the actual values through the transfer function to see the actual frequency responses.

I don't think I ever told Kieth, but, using these programs I designed the filters for Steve StCroix (Marshal) in his awesome BBD based Marshal Time Modulator. Steve and I remained friends for a long time.

The filter section is also where we did pre-emphesis (HF boost) and de-emphesis (HF cut) to reduce high frequency artifacts that result from the quantization of audio.

compressor expander and limiter.
10 bits was not enough. To improve the dynamic range, audio was compressed going in and expanded coming out. This is a huge help.

To do this I used a part from Signetics called the NE571. It could do 2:1 compression and expansion. I got to know the designer of this IC, Craig Todd. So, if 2:1 is good, more is better, right? There is a circuit in the path to the detector consisting of 2 transistors and 3 resistors that simulates back to back zener diodes. This addition allows greater than 2:1 compression and expansion.

We wanted a limiter too, so, there is an added detector that creates a fairly brick wall limiter. Having a limiter in the loop is really fun when doing echo and layering phrases. It allows the musician to replace the existing repeating phrase with a new phrase by playing the new phrase loud.

This section determined a lot about the sound of the unit. If you went hard in to limiting, a lot of stuff happened in terms of messing with the dynamics of the audio. The expander could decompress the compressor, but, not the limiter.

Memory
22 pin 4k DRAMs. These were very new, and expensive. I chose the 22 pin because it was created by Intel and had other sources coming on. There was a 16 pin Mostek design that

ultimately won the battle between 16 and 22 pin parts due its smaller package. The 22 pin gave us a competitive reliable sources of parts at the time.
Designing with these parts was difficult. They had to have a 12 volt supply. The chip enable (CE) signal had to swing the entire 12 volts at very fast rise and fall times while the ICs seemed to load this CE pin with a huge amount of capacitance.
There are 2 versions of the CE driver located on the main PCB, one was an IC, the other was a discrete design of mine.

Audio input output circuitry which included mixing, regeneration, level matching.
The main unique thing here is the use of a single control for both input and output levels. I thought it was a great idea at the time, but, the world seems to prefer separate input and output level controls. Regeneration is turned off when bypassed.

Digital control circuitry including a variable frequency high speed clock
Changing delay time, in real time, was done by varying the high speed master clock that drove the entire digital system. In order to be able to do a 4 to 1 sweep, which we felt was necessary for flanging, I needed a high speed clock that was controllable by a current from analog delay time controls. It needed to operate in the MHz region, I couldn't find anything off the shelf at the time, so, I had to make my own. My solution was driving 74LS221 one shots with a current mirror. The one shots are wired back to back to create a square wave oscillator with decent linearity over the range I needed. I thought this was a creative solution.
I won't go over the digital section in detail. One thing that is not obvious is that the memory cycles have refresh cycles in addition the normal memory read write cycles. There was something in 3's. Perhaps it was 2 refresh cycles for every read/write cycle. So, I think it accesses every third location. I remember having to deal with an issue when 3 memory boards were selected. Memory cells required refreshing every 2 ms. worst case.

modulation of the delay with a low frequency oscillator.
This portion was probably taken largely from the Auto Flanger. The output of this circuit is a current that goes to the Variable high speed clock which resides with all of the digital circuitry. Driving with a current helped minimized noise in the frequency control. What is not obvious is that the transistor driving this current to the digital section is actually an exponential amplifier. The base of Q4, a 2N5172, is being driven by a voltage from a resistor divider with a fairly low resistance, 1k ohm. The emitter is effectively tied to a reference, so, when the voltage on the base varies, the current is an exponential of the voltage. This is the kind of circuit that Keith Barr was brilliant at. I learned a number of circuit design chops from Keith.

AC power supply
Lots of voltages.+/- 15 for analog, +12 for DRAMs, +5 for digital, -5 for DRAMS, +/- 7.5 for CMOS switches. Symmetry of the +/- 15 was important, hence the trim pot. The AC transformer was custom.

Biggest design weekness.
By far the greatest source of problems associated with the 113 was its connector system for connecting the boards together. Pins from the main PCB connected to sockets on the plugin boards. Socket connections were what is known as a tuning fork attempting to connect to opposing sides of the pins. These tuning forks would open up over time and not grip the pins and create intermittent connections.

The best fix for this I heard about was from a service tech on the west coast. He would disassemble the boards, remove the connectors and replace them with wire soldered to both boards. This meant that the unit could no longer be serviced by board swapping. But that didn't matter because the biggest issue was fixed. This would have made it more road worthy.

Also, foam rubber was stuck to the inside of the cover to hold the boards. This foam rubber deteriorates over time and becomes useless allowing the PCBs to move more further exacerbating the connector issue.

A less prominent problem was, one of the regulators was mounted on the rear of the chassis. The connector used to connect to this TO-220 regulator didn't grip the pins well. I would recommend elimination the connector and solder the wires directly to the IC pins with shrink tubing over the connection.

Other than those issues the design was perfect :-)

Reflections upon looking back over almost 50 years.

In reviewing the schematics and reflecting back on the whole project a few things come to mind.

1) A very minimalist design. There are no extra parts. For example, it is common practice to put ceramic capacitor on the power pins to digital integrated circuits. I didn't put them in there. Just a few strategic bypass caps as needed.
2) It is a very creative design at almost every section. It was a huge amount of work.
3) What company besides Keith Barr's MXR would allow a self trained engineer in his early 20s take on a project like this? Keith trusted me.
4) What a fun product. It made you want to play. It had a lot of depth to explore. One could use to simply add some space and chorus to an instrument. It could also do looping stuff which can occupy a musician for hours. And use modulation to ridiculous levels to totally shred the audio beyond recognition.
5) The repeat hold feature may have made it the first digital looper.
6) most of the "sound" of the unit is in the analog domain. With the amount of filtering and dynamics manipulation going on, it is hardly a straight wire with delay. This is evident when using it for echo and turning up the Regen and hearing the repeats decay and the artifacts accumulate. I had to make many compromises and trade-offs in the design and I tried to make the artifacts as musical as possible. I used my ears a lot.

This product was career defining for me. It was used by many of my favorite musicians, like Jaco Pastorius who played it like a musical instrument. I've been told that I changed the music industry, which seems like a bit much. Its success for MXR gave me a lot of freedom. It allowed me to spend 2 years figuring out how to make a Pitch Transposer, another career defining product for me.

I am very blown away by the musicians that used it including some of my favorites of the time including: Chick Corea, Jaco Pastorius, Jean Luc Ponty, Goerge Duke, David Gilmour, Roger Waters. This is hardly a complete list.

I had a jaw dropping moment with the Digital Delay in 2006. I purchased a multi-CD Weather Report compilation. It included a DVD with a live concert from Frankfort Germany. Jaco did a long solo which would best be described as "Solo for Fender Bass and MXR Digital Delay". I hope it doesn’t sound arrogant, but he did things with it that only an MXR Digital Delay could do. When I was reading the credits, I realized that the when the concert happened, I was only 25 years old. That's what made my jaw drop. How many 25 year old engineers have been able to take advantage of the opportunities I had. I have had a great career.

I was not alone. I received help from many at MXR including Keith Barr, Richard Neatrour, Dave Yeager, John Porubeck and others in engineering. Many others helped get this thing in to production in a factory built to make pedal effects including Terry Sherwood, Michael Laiacona, Phil Betette, John Langlois, Ron Wilkerson.
We had quite a ride.