The analysis of Bucket Brigade Devices (BBD), early forms of sampling

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Bucket Brigade Devices (BBD) are used mainly in analogue guitar pedals to create effects like echo, delay, flanger, and chorus, and have a following of musicians who love their particular sound characteristics. BBDs are also related to analogue to digital converters (ADC) in that they sample the incoming signal but do not quantize it like ADCs do. Studying the BBD architecture and analysing two BBD guitar pedals, the TC Electronic Echobrain, and the Boss DM-2W sound signature response and hardware gives an application of sampling for students learning about analogue to digital conversion ADCs and a better understanding on how to closely model delay pedal effects in a simulation.

Bucket Brigade Device | Circuit analysis

1. Introduction

A. BBD chip structure. Developed in 1968 to 1969 by F. Sangster and K. Teer in the Philips Research Labs (1), the Bucket Brigade Device (BBD) is an analog chip that creates a signal delay effect commonly used in Delay, Flanger, Chorus and Reverb pedals.

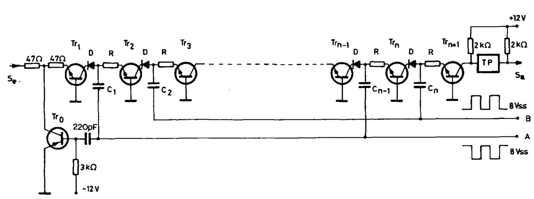

Fig. 1. Represents an original version of a BB Chip (2)

BBD's use the bucket-to-bucket concept, where water is passed in a human chain from one point to another. In this case however it is a voltage signal that is passed from capacitor-to-capacitor N times using a switch. A signal would be sampled and stored in the first capacitor in the chain. Then based on Clock A or B in Figure.2 the switch will close, allowing the passage of the signal to the next capacitor. The clocks are phase shifted by pi (inverted), allowing one switch to be open while the other is closed.

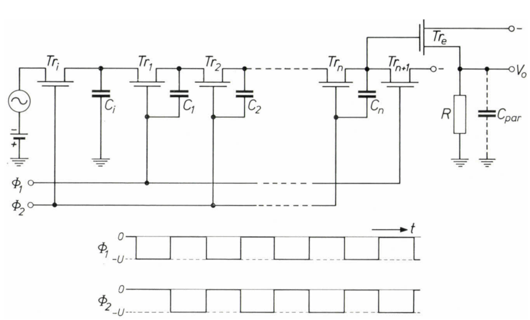

Fig. 2. Standard MOSFET BBD circuit (3)

B. BBD chip circuit architecture. In order to better understand the architechture of BBD effect pedals, we shall take a bried look at each component to see what their purposes are before going into a more in depth analysis of its hardware.

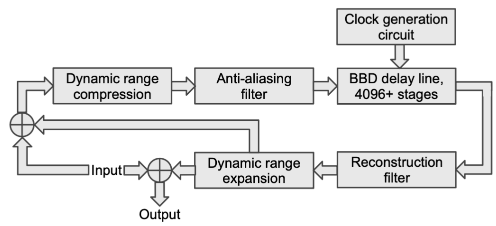

Fig. 3. Simple BBD circuit Architecture (4)

B.1. Compressor and Expander. The compressor will take the input and reduce its dynamic range to accommodate for the BBD's max voltage levels, or else we will have signal distortion from over-saturation of the capacitors in the delay line. Compared to the saturation achieved on tape delay machines it is not as desirable as an effect.

To bring back the dynamic range of the original signal before being recombined at the output, the sampled signal is expanded to match the original dynamic range and when combined with the original signal will be heard.

B.2. Anti-Aliasing Filter and Reconstruction filter. Let f_s be the sampling frequency of the BBD, which is also the clock frequency. Following this equation $TimeDelay = \frac{N}{(2f_s)}$(4), we can see that at higher levels of capacitors and lower clock frequencies, we can get very long time delays, however with that we run into the problem of Aliasing when $f_{signal} > \frac{f_s}{2}$, we would thus apply an AA Low Pass Filter to the signal with a cut off frequency (f_c) at -3dB before the Nyquist frequency of the clock, as such we will attenuate high frequencies of the signal at long delay times. However, for quick delay time effects like flanger and choruses, there is no need for a AA Filter since a flanger created with a ten millisecond delay time and 512 capacitors has a 25 600 Hz for a sampling rate, with a Nyquist frequency of 12 800 Hz, creating a significantly less amount of Aliasing. Hence for longer sample periods, an AA filter is needed.

To reconvert from discrete back to a continuous signal upon exiting the BBD chip, a Reconstruction filter is used to smooth the signal out.

What we will see later on is that most BBD chip architectures will use Sallen-Key AA and Reconstruction filters of 2^{nd} and 3^{rd} order. The common Sallen-Key architecture

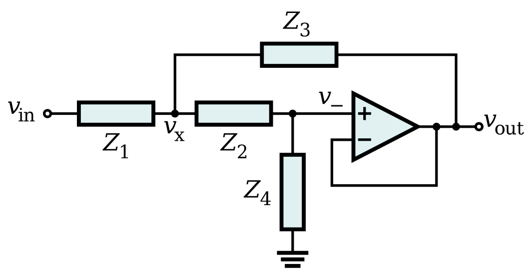

Fig. 4. Generic 2nd order Sallen-Key filter using an OP-AMP (5). Modules Z1 and Z2 would be resistors, and Z3 and Z4 would be capacitors when implementing a Low-Pass filter and vise-versa for a High-Pass filter. This filter designer would be thus used in circuits to implement AA and Reconstruction filters as it is efficient and has good levels of attenuation.

uses a series of RC circuits in order to increase with every degree the precision and slop of the filter. In Figure.4's case it would be groups Z1,Z3 and Z2,Z4 which are RC pairs in series. RC circuits are essentially natural filters. As the signal crosses the resistor, the capacitor holds the signal sample, delaying it for an x period of time only to add it back to the circuit afterwards. At a lower signal frequencies, the delayed signal will add back at a time period where the signal has not greatly changed, only increasing the signals amplitude. At higher frequencies, the delayed signal will instead subtract as there has been a greater change in the signal. This results in allowing only lower frequencies in the signal to pass while the higher ones are canceled out.

2. Hardware Analysis

Now that we have an in-depth understanding of how BBDs work, we shall see how the chip and its circuit are applied by analyzing the circuitry of the Boss DM-2 Delay effect pedal as its schematics have been made public.

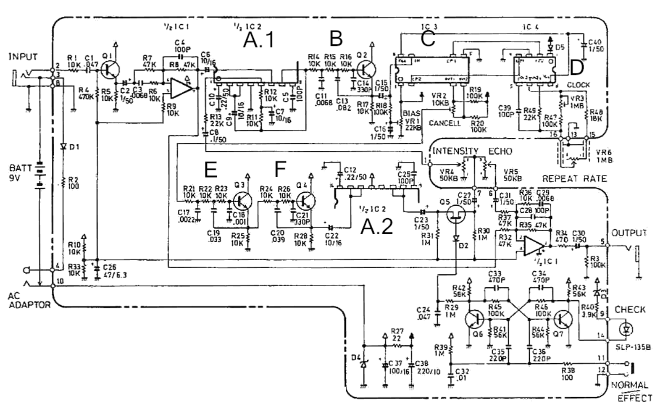

Fig. 5. Boss DM-2 Delay guitar pedal schematic diagram(6):
A.1 is the Compressor Component of the NE570N Compander.
A.2 is the Expander section of the NE570N Compander.
B a 3^{rd} order Sallen-Key Low Pass Filter.
C is the MN3005 or MN3102 BBD with 4096 stages.
D is the MN3101 or MN3012 BBD clock generator.
E and D are a series of 3^{rd} and 2^{nd} order Sallen-Key Reconstruction filters.
The MN Series (7) was manufactured by Panasonic/Matsushita and was the most commonly used BBD chip in guitar pedals at the time.

A. Boss DM-2. The Boss DM-2W is a guitar delay effect pedal that was widely used in the late 1900's developing a cult following due to it adding a warm color to the delayed sound. It has been also widely copied and emulated in software making it a good starting point in our analysis of BBD guitar devices.

A.1. BBD chip. Next, the signal is passed through the BBD chip (MN3xxx) who's clock rate (sampling rate) is determined by the clock Generator (MN310x). Looking at section D of 5 and co-referencing with its data-sheet (7), we can see that with the capacitor $C39 = 100pF$ and the resistor $R49 = 22k$ going into ports 5 and 7 of the clock generator, the clock output frequency (CP1, CP2) can range in between 2.6 $220kHz$. With the BBD having 4096 delay stages and a delay time between 20-300ms, the range of the clock delay time is:

$$DelayTime(s) = \frac{N_{stages}}{2 * f_{cp}} \longrightarrow f_{cp} = \frac{N_{stages}}{2 * DelayTime}$$

$$f_{cp_{min}} = \frac{4096}{2 * 0.020} = 102.4kHz$$

$$f_{cp_{max}} = \frac{4096}{2 * 0.3} = 6.82kHz$$

With a range of 6.82 102.4 kHz, we can tell that there is a need of an Anti Aliasing filter to reduce any noticeable amount of Aliasing from frequencies that are beyond the Nyquist frequency ($f_{Nyquist_1} = 6.82kHz/2 = 3.41kHz$, and $f_{Nyquist_2} = 102.4kHz/2 = 51.2kHz$).

A.2. Filters. In order to reduce aliasing that may occur due to frequencies being above half of the clock frequency(Nyquist frequency), we see in Figure.5.(B,E,D) the signal is first passed through a 3^{rd} order Sallen-Key Low Pass Filter before the BBD chip and then a 3^{rd} order Sallen-Key Reconstruction Filter in series with another 2^{nd} order Sallen-Key Reconstruction Filter. In order to see what effect these filters have on the signal, we are going to use the standard Sallen-Key filter equation that models the filter using an OP-AMP (operational amplifier) instead of a transistor in the filters of Figure.5. This is due to the similarity in the frequency response between the two variations with more reliable references for the OP-AMP version making it simpler to model.

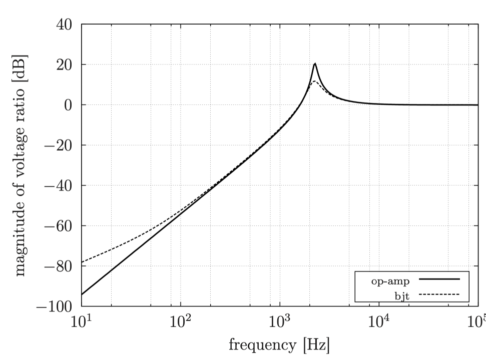

Fig. 6. Comparison between OP-AMP and BJT High-Pass filters (8).

In this case, Figure.6 is comparing a High-Pass op-amp and BJT(explain this) based Sallen-Key filter. As it is just the inverse of a Low-Pass filter (swapping the capacitors with the resistors in the circuit diagram) this graph shows that an OP-AMP based Sallen-Key filter will have greater attenuation and a greater peak at the center frequency making the preferred choice for modeling the Sallen-Key architecture. To model the 3rd order Low-Pass filter, the equation used is:

$$\frac{V_{out}(s)}{V_{in}(s)} = \frac{b_0}{a_3 s^3 + a_2 s^2 + a_1 s + a_0}$$
$$b_0 = (C_1 C_2 C_3 R_1 R_2 R_3)^{-1}$$
$$a_0 = (C_1 C_2 C_3 R_1 R_2 R_3)^{-1}$$
$$a_1 = (C_1 R_1)^{-1} + (C_1 R_2)^{-1} + (C_2 R_3)^{-1} + (C_2 R_2)^{-1}$$
$$a_2 = (C_2 C_3 R_2 R_3)^{-1} + (C_1 C_2 R_2 R_3)^{-1} + (C_1 C_2 R_1 R_3)^{-1} + (C_1 C_2 R_1 R_2)^{-1}$$
$$a_3 = 1$$

Using these coefficients from a Sallen-Key filter used in a Boss DW-2 (6):

$$C_1 = 0.0068\mu F; C_2 = 0.082\mu F; C_3 = 330pF;$$
$$R1 = R2 = R3 = 10k\Omega$$

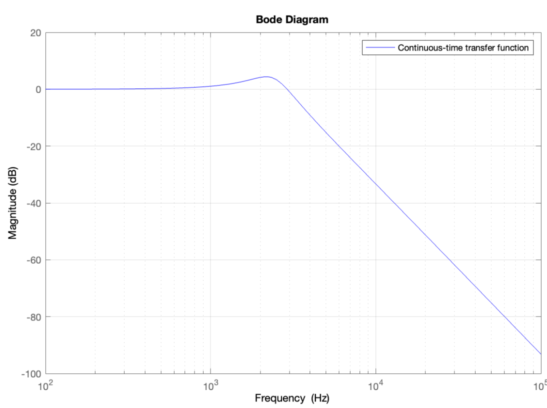

Fig. 7. 3rd order Sallen-Key Low-Pass filter designed in Matlab using the tf function (9) with the coefficients mentioned above.

Similarly, for the 2nd order Low-Pass filter, the equation is:

$$\frac{V_{out}(s)}{V_{in}(s)} = \frac{b_0}{a_2 s^2 + a_1 s + a_0}$$
$$b_0 = (C_1 C_2 R_1 R_2)^{-1}$$
$$a_0 = (C_1 C_2 R_1 R_2)^{-1}$$
$$a_1 = (C_1 R_1)^{-1} + (C_1 R_2)^{-1}$$
$$a_2 = 1$$

With the coefficients:

$$C_1 = 0.0039\mu F; C_2 = 330pF; R1 = R2 = 10k\Omega$$

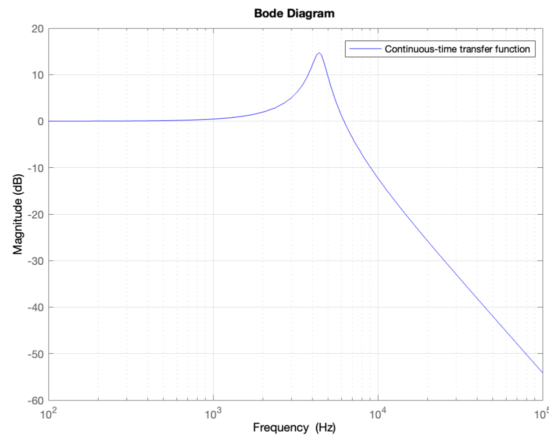

Fig. 8. 2nd order Sallen-Key Low Pass filter designed in Matlab using the tf function (9).

A.3. Compander. In Figure.5.A we see that it is first passed through the compressor component of the NE570N Compander. As we have described earlier is used to reduce the dynamic range of the incoming signal so as to no overload the capacitors in the BBD chip during the sampling period so as to prevent potential harmonic distortion. It makes sure to maintain the signal to noise ration of 60dB. Looking at it's data-sheet(10) the NE570 will attenuate the signal by half going it at a decibel range of -80 to 20 dB. It will then expand it back after before summing the delayed signal to the input signal that is not only going back into the circuit loop but also exiting from the output of the pedal.

3. Software Analysis

Now that we have understood how the analogue variants function, we can proceed to model them. In this section we will talk about our methods of modeling the Boss DM-2 guitar pedal

A. FAUST. To model the DM-2 we used the programming language FAUST (Functional Audio Stream), a functional programming language that focuses of sound processing. It is meant to be simple and intuitive to learn, also offering a block diagram to help visualize the flow of the signal through the modules that make up the software.

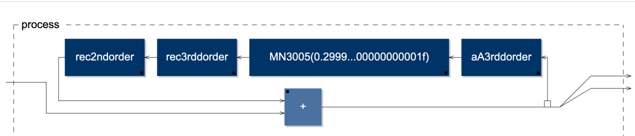

Fig. 9. The Block diagram for the FAUST Code.
Here the input is being added to the feed-back signal that is passed through (from right to left in the loop) the 3rd order AA filter, the delay line that mimics the MN3005 chip, the 3rd order Reconstruction filter and finally the 2nd order Reconstruction filter.

B. Filter modeling. To model the filters, we took the transfer function obtained in Matlab and applied a bilinear transformation(11) on it using the c2d function (9) to convert from a continuous to a discreet transfer function (s plane to z plane). Then, the coefficients are implement in a Infinite Impulse Response Filter(IIR) (12) in our FAUST code. An Infinite Impulse Response Filter essentially feeds back the output to the input.

$$y(n) = \frac{1}{a_0}(b_0 x[n] + ... + b_K x[n-k] - a_1 y[n-1] + ... + a_i y[n-i])$$

$$H(z) = \frac{\sum_{k=0}^{K} b_k z^{-k}}{\sum_{i=0}^{I} a_i z^{-i}} = \frac{\sum_{k=0}^{K} b_k z^{-k}}{1 + \sum_{i=1}^{I} a_i z^{-i}}$$

Fig. 10. This is the IIR's differential equation where x[n] in the input and y[n] is the output and it's transfer function where $a_0 = 1$ usually. From the transfer function the "a" and "b" coefficients are used when creating a digital IIR filter. "a" are the feedback coefficients and "b" are the feedforward coefficients.

Here are the results of the discretized filter:

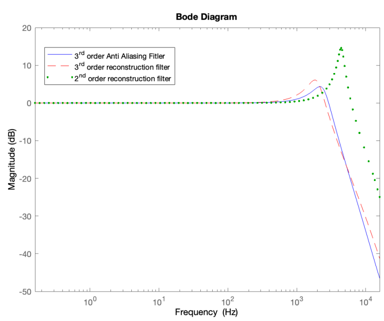

Fig. 11. Discretized filters model from the Boss DM-2 guitar pedal. They are implementations of 3rd and 2nd order Sallen-Key Filters

C. BBD delay modeling. To design the delay line, it can be as simple as delaying the signal for an x amount of time and then add it back to the new non-delayed input. Then to prevent any noticeable clicks when changing the sample rate of the delay line we can interpolate the delayed signal. This has an added benefit of modeling the time stretching effect heard in BBD pedals when changing the delay time.

The a simple method would be to linearly interpolate the value using the equation:

$$y[n + \nu] = (1 - \nu)y[n] + \nu y(n + 1)$$

where $\nu = [0, 1]$ and can derive a value between two sample points. Lagrange interpolation can also be used for better results (11).

D. Compander. The main purpose of a compander was to prevent overloading of the BBD chip while preserving its dynamic range from input to output. This however is not required in software as the input is generally normalized and the delay line software does not have voltage limitations like the capacitors in the BBD chip do. Therefor it has little difference in the frequency response of the simulation to have a compander implemented or not.

4. Results

By simply modeling the filters found in the circuit diagram of Figure.5 using the determined equations mentioned in section 2.B.2, we are able to have a similar sound profile to the C++ model(4). The C++ model differs from the FAUST model in its use of higher order polynomials to increase signal stability through the delay chain and the rounding of its peak in its frequency response as seen in Figure.12. The lack of a compressor and expander in the FAUST code also shows minimal change to the sound profile in comparison to the C++ model. FAUST handles the audio routing and processing in the background leaving audio manipulation to the user to define with their code. As such, the FAUST code is simpler to understand compared to the C++ code. The block diagram generated by the FAUST Code also helps to understand what is being done in a simple and detailed way.

By passing white noise for a split second through the DM-2W, Echobrain, FAUST code, and C++ code, we took only the delayed portion and plotted its frequency response out in Figure.12. We see that as mentioned above there is stronger peaks in the FAUST code due to its lower order polynomials. The presence of a compander in the C++ code creates harmonic distortion in the -60 and -80 dB range which as a result is unperceived when listening back to the audio. The DM-2W and the Echobtain both have peaks in their frequency response's attenuated region. This is due to Aliasing, however the AA filter in the pedals is preventing it from being noticeable. As such we have left it out in the modeling of our FAUST code.

5. Conclusion

By researching the BBD architecture, and analyzing the original Boss DM-2, we have deepened our knowledge of the sampling portion of the ADC process and learned how to model delay pedals. It was observed that the filters on analogue delay pedals can be modeled using the Sallen-Key low pass filter transfer functions, and that the delay line and it's time stretching effect can simply be simulated using the interpolation method. While an important part of a BBD pedal, a Compander circuit model was not needed in order to reproduce the frequency response of the pedal. However, for accuracy of reproduction a Compander would be required.

To better understand the components of a BBD guitar pedal, more research is required to determine the cause for the lower cut-off frequency of the analyzed BBD pedals in Figure.12. This would imply measuring the components of the BBD pedals and determine which are part of the filter circuits model them using the methods discussed in this paper. Furthermore, research into modeling companders is required as there is scare documentation on the subject.

One topic not touched upon is the mimicking of the aliasing found in the BBD chips due to it's sampling rate by down-sampling the signal at the input and up-sample is at the output.

All code mentioned in this paper can be found in this repository: https://gitlab.com/Maxw3llGM/bbd

6. Bibliography

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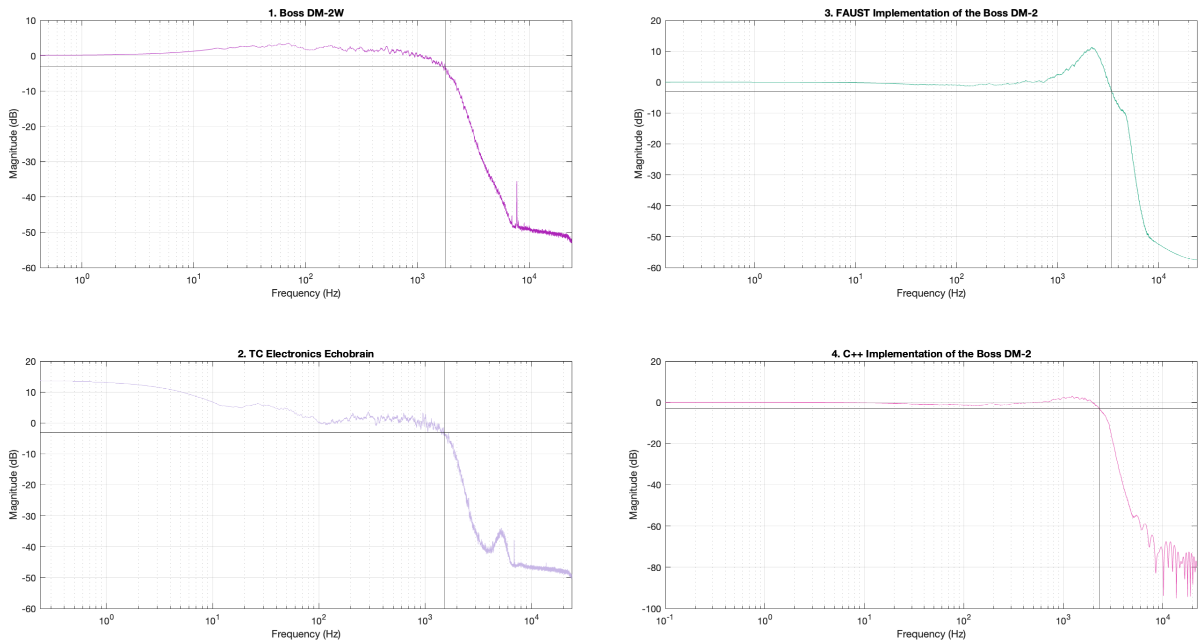

Fig. 12. We obtained these frequency responses by passing white noise through each device.
1. Boss DM-2W with a cut-off frequency of 1768.82 Hz
2. TC Electronics Echobrain with a cut-off frequency of 1516.73 Hz
3. FAUST Implementation of the Boss DM-2 with a cut-off frequency of 3437.51 Hz
4. C++ Implementation of the Boss DM-2 with a cut-off frequency of 2297.89 Hz

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