THE ANALOG CHORUS, ORIGINS OF THE EFFECT, WORKING AND STUDY OF THE BOSS CE-2

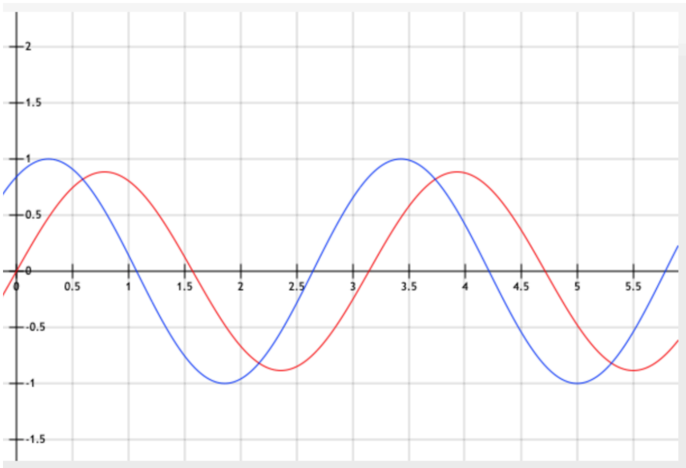
By Alexandre Antoine on 6 August 2021

In this new article, we're going to talk about the working principle of a modulation effect that is very popular with guitarists who love a soaring atmosphere, the chorus! It's a modulation effect based on a technology similar to the analog delay, but which presents some particularities. So we're going to look at the origins of the chorus and how it works. Then we will study the circuit of the Boss CE-2, a very famous analog chorus pedal.

FOLLOW OUR TONE QUEST !

WHAT IS THE CHORUS EFFECT ?

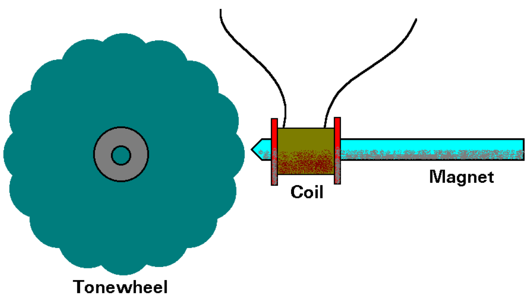
The name chorus obviously refers to a group of vocalists singing together. The effect is designed to reproduce the idea that several instruments are playing the same musical piece with a slight delay, and with slight variations of pitch. This gives the impression of a massive sound coming from a choir, hence the name! Based on modulation, but also on delay, we can illustrate the effect like this:



In blue, the original guitar signal, without effect. This signal is duplicated, delayed and slightly pitch shifted. The two signals mixed together form the chorus effect.

ORIGINS OF THE EFFECT, WORKING OF THE MECHANICAL CHORUS GENERATOR OF HAMMOND

The first chorus appears in the fifties on Hammond organs. It was a mechanical system, as a result of criticisms about the too flat sound and lacking realism that the organs produced.

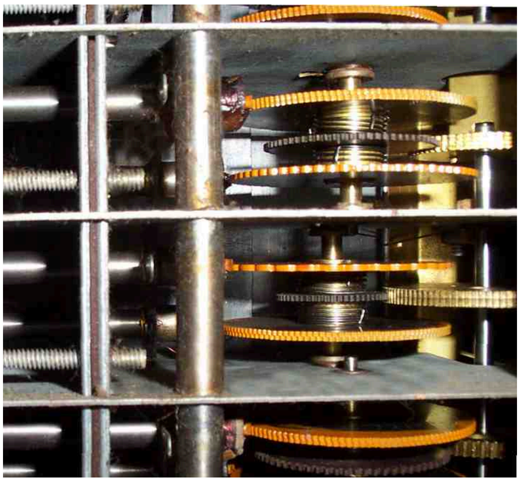


Simplified diagram of the mechanical chorus generator on the Hammond organ. Source : theaterorgans.com.

bar induces a voltage in the coil.

By adjusting the rotation speed of the wheel, we can generate an alternating voltage at a certain frequency. In fact, there are a number of lobes on the wheel. Depending on the number of revolutions the wheel makes, we can for example pass 440 lobes per second, which gives a sound at 440Hz.

In fact, the organ is composed of several wheels stacked together, but with one lobe added or removed, producing several notes, very close to each other but slightly pitch detuned. Hence the chorus effect. I let you imagine the weight and the size of the instrument!



Tonewheels stacked with driving system. Source : theaterorgans.com.

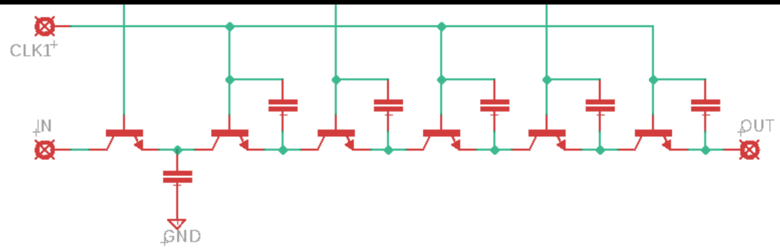
The result is a nice sinusoidal sound generator. Mixed with the direct signal of the instrument, it produces a beautiful chorus effect. More information about the Hammond tonewheel organ.

THE ANALOG CHORUS EFFECT, AND THE WORKING OF THE BOSS CE-2 PEDAL

In the 70's, due to the success of the Roland JC-120 amp, the Jazz Chorus, the brand commercialized the very first chorus in pedal format, the Boss CE-1. It includes the same electronic circuit as the JC-120.

The Boss CE-1, the most vintage chorus pedal.

Here, the delay is created with a BBD chip. This little chip, which we have already talked about here, was the first solution to offer delays in pedal format. It is still used today in analog chorus pedals. It gives a very warm and vintage sound, and you can easily make it saturate by adjusting its input gain.

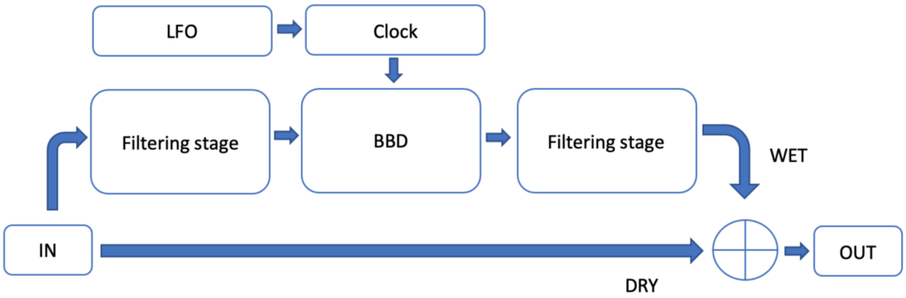


Classic topology of a BBD chip.

Let's just remember that a BBD contains a series of capacitors and transistors, driven by an external clock. Two clock signals work alternately and command the opening and closing of the transistors. The signal is then progressively transmitted from one capacitor to another, and thus takes a certain time to go through the chip, which creates a delay. With the chip that we use, we can generate delays up to 50ms, which is more than enough for the chorus, but we can put them in series to create much longer delays.

We will now look at the rest of the circuit by studying the schematic of the Boss CE-2, a famous analog chorus pedal, to understand how to modulate the pitch of the delayed signal.

GLOBAL STRUCTURE OF THE CIRCUIT



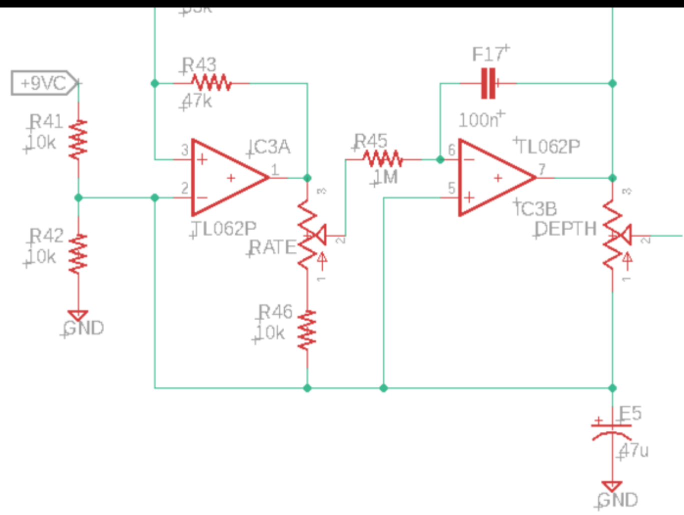
As we have seen above, the signal is first splitted. In one hand, the DRY signal goes directly to the output, and on the other hand, the WET signal goes into a first filtering stage. This filtered signal then goes into the BBD to be delayed. Note that the BBD is driven by an external clock, which is itself driven by an LFO, to produce pitch variations. Then the delayed signal goes through a new filtering stage before being mixed with the DRY, this is the signal that we hear at the output of the pedal.

But what is the LFO ?

THE LOW FREQUENCY OSCILLATOR

The LFO, or Low Frequency Oscillator, is the heart of the chorus effect. As its name suggests, it is a low frequency wave generator whose shape can be chosen. For a chorus, we go from mHz to a few Hz only, it is enough for this application. The chorus is thus created by the slow modulation of the delay time. And this modulation comes from the LFO!

This is because the LFO varies, the delay time produced in the BBD is constantly increasing and decreasing and as a result, the pitch of the delayed audio signal is slightly pitched up and down.



Schematic of the LFO circuit, it is the Boss CE-2 one.

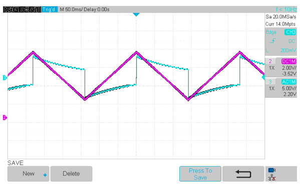
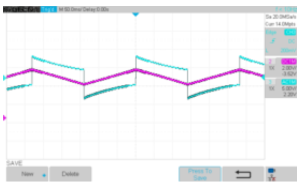
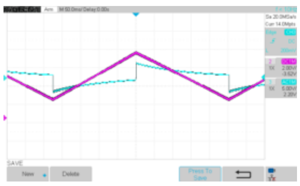
The LFO circuit implements a dual op-amp. The first op-amp is a threshold detector in a Trigger-Schmitt configuration with positive feedback. This first part of the circuit generates a square wave signal, whose frequency varies with the RATE pot. You can customize the LFO to obtain a much faster maximum speed by decreasing the R45 resistor, for you vibrato lovers…

The second op-amp is an integrator circuit, which gives …. a trianglular waveform at the output. This time the DEPTH pot adjust the amplitude of the signal, in other words the depth of the effect.

We have been studied several chorus pedal circuits (including the Boss CE-2), and the majority of analog LFOs provide triangular waveforms. The pitch changes are therefore faster and more sudden than with a sinusoidal LFO, but sinusoid is more complicated to produce with analog circuits…

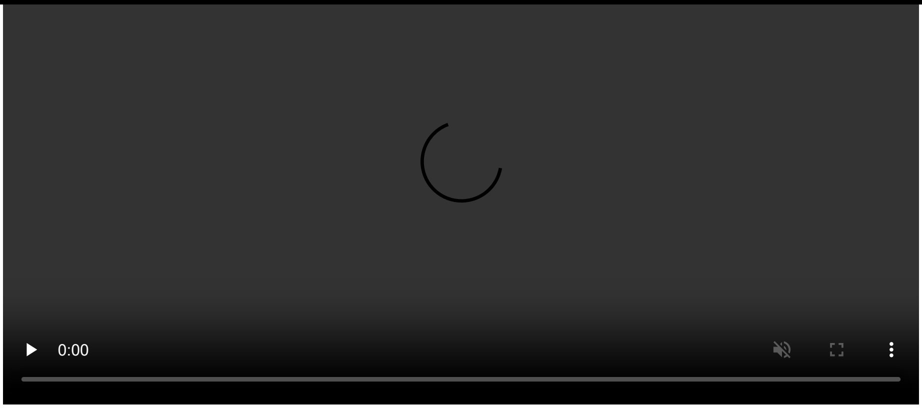
SOME OSCILLOSCOPE MEASUREMENTS

The measurements below correspond to the two outputs of the LFO. The blue signal corresponds to the square signal taken at the output of the RATE potentiometer, while the purple corresponds to the triangular signal at the output of the DEPTH potentiometer.

- ON THE LEFT A FIRST MEASUREMENT WITH THE 2 POTS AT THE MAXIMUM, THE OSCILLATION FREQUENCY IS 3.5HZ.
- IN THE MIDDLE, THE DEPTH IS LOWERED, THE SIGNAL AMPLITUDE VARIES, AND THE FREQUENCY REMAINS THE SAME.
- ON THE RIGHT, THE RATE IS LOWERED AND THE FREQUENCY DECREASES, UNTIL ABOUT 300 MHZ, THE AMPLITUDE REMAINS MAXIMUM.

In summary, we have a triangular signal that drives the clock, which has the effect of modulating the two clock signals controlling the BBD. Thus, the delay time created by the BBD is always changing. You can check below the effect of the modulation of the clock by the LFO:



WE VARY THE RATE POT OF THE LOW FREQUENCY OSCILLATOR, THE CLOCK SIGNAL IS MODULATED!

So as we said, this modulation of the clock signal produces a variation of the delay time in the BBD, and therefore a pitch variation of the signal! This modulation can also be found on an analog delay pedal. Take your favorite analog delay pedal and slowly vary the delay time while you play – not easy I agree – but you will hear a modulation similar to a chorus! And for those who are interested, we actually have a delay pedal that adds modulation to the delayed signal ! It's over here:

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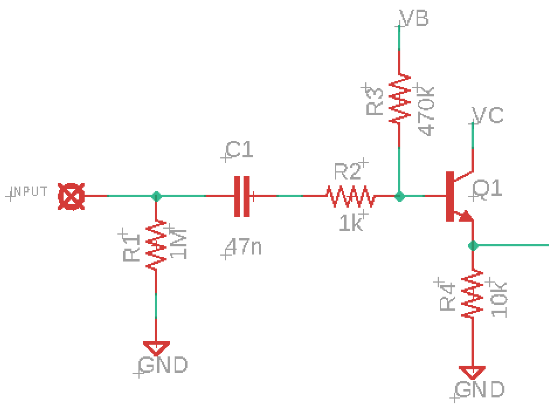
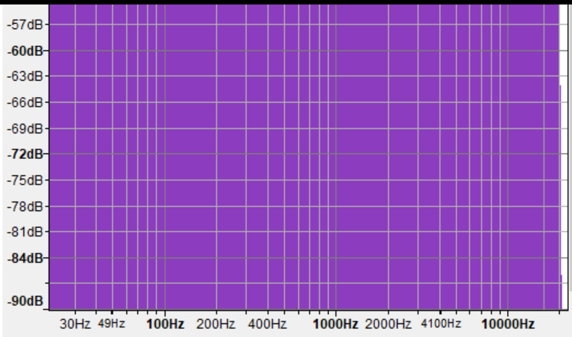
Now we will discuss about the other analog filtering stages in the Boss CE-2 pedal circuit.

INPUT BUFFER

The first step is to buffer the input signal in order to preserve its audio quality. We have already talked about this in the Ego Driver article, but it is always good to remember its purpose.

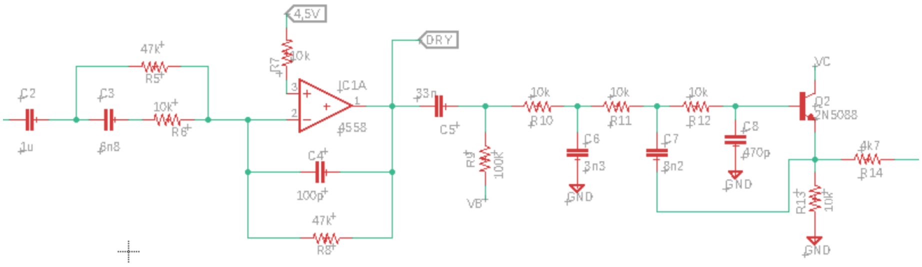


This circuit preserves the harmonic content of the input signal, so its spectrum is flat in the bandwidth:

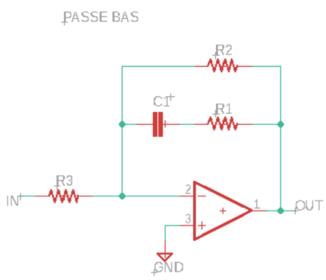


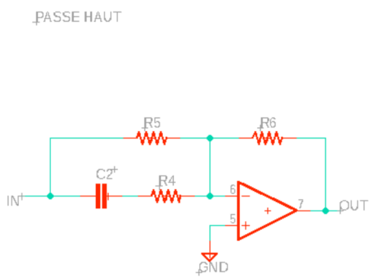
INPUT FILTERING STAGE

Once the signal is adapted to the correct impedance, we go to the following filtering stages:

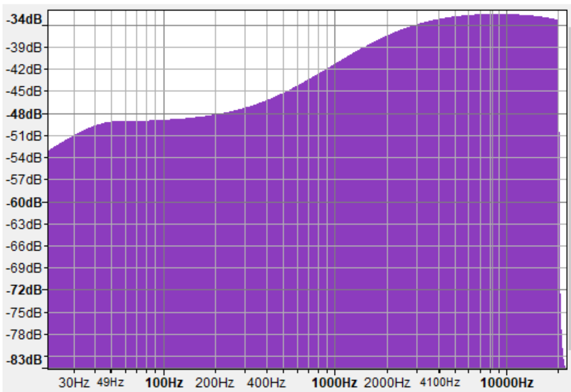


The first filter, called pre-emphasis filter (just before C5) is a filter that boosts the high frequencies. We use a Shelving filter topology for this:



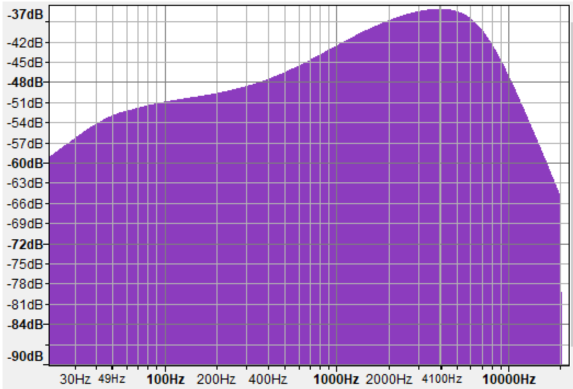


The idea is the following : By applying a boost of the high frequencies and an identical attenuation to the DRY signal at the output, we find the same signal as at the input, which you may think is not very interesting. On the other hand, by applying this same attenuation to the WET signal at the output of the BBD, we remove all the potential noise in the high frequencies acquired in the BBD! We also find a complementary filter for the attenuation of high frequencies at the output stage of the circuit.



Here the amplitude of the high frequencies is increased (from 1khz).

know that it is normally used when a signal is sampled by an analog to digital converter. But then, what is the interest in our case? Well, it turns out that the BBD is not really analog. The BBD chip samples the signal by storing charges (discrete values) in its capacitors. To avoid that the high harmonic content of the signal disturb the BBD and create spectral aliasing, they are suppressed with this low pass filter.

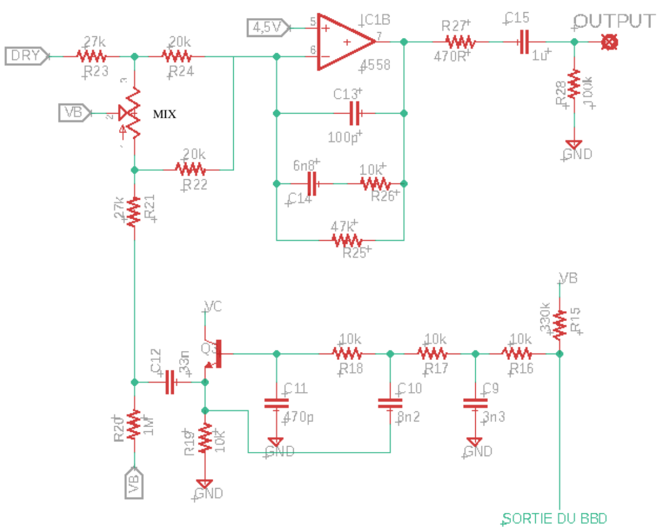


The low pass filter cuts around 5khz.

THE BBD : ANALOG OR DIGITAL?

Be careful, if the BBD is not completely analog, it is not 100% digital either, but rather analog-voiced. Because a digital signal is also quantized. That is to say that each sample cannot take the value it wants. It can only be rounded to certain levels, which are usually associated with numbers that can be processed by a processor. In the BBD this is not the case, our samples can have any value, they are not quantized. So the BBD is analog-voiced.

OUTPUT FILTERING STAGE



So, our signal is now delayed and modulated by the LFO! There is one last filtering stage. As we explained before, the BBD is not 100% analog because it creates samples of our signal. So we have a sampled signal at the output of the BBD, that we need to convert back to analog. We use a low pass filter again (similar to the anti-aliasing filter) to reconstruct a "smooth" analog signal from the sampled signal. By adjusting the capacitors of this filter, and also the high-pass filter formed by R20 and C12, we can modify the tone of the circuit, and thus propose several colors of effect: we can go from a clear and bight chorus to a dirtier, darker chorus.

The last filter is called de-emphasis filter. Complementary to the boost filter seen above, it brings the signal back to its original frequency response by attenuating the high frequencies. And also mix the DRY and WET signals.

CONCLUSION

then again, analog is way more charming… I have the impression that a FX Teacher chorus kit would tempt you, am I wrong?

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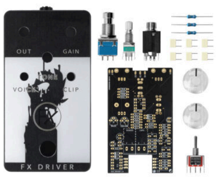

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