Oakley Sound Systems

DCR320 – Stereo Chorus

User Manual

V1.2

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Introduction

This is the User Manual for the DCR320 stereo chorus rack module from Oakley Sound. This document contains an overview of the unit and goes into some detail regarding the operation of the module. It also contains the calibration instructions.

For the Builder's Guide, which contains a basic introduction to the circuit board and a full parts list for the components needed to populate the board, please visit the main project webpage at:

http://www.oakleysound.com/DCR320.htm

For general information regarding where to get parts and suggested part numbers please see our useful Parts Guide at the project webpage or http://www.oakleysound.com/parts.pdf.

For general information on how to build our modules, including circuit board population, mounting front panel components and making up board interconnects please see our generic Construction Guide at the project webpage or http://www.oakleysound.com/construct.pdf.

The Oakley Sound DCR320

The Oakley DCR320 is a stereo ensemble unit designed to mimic the behaviour of a well known stereo chorus rack effect used in countless audio productions since the early 1980s. The original unit, and its various clones, have only four preset buttons on the front panel which limited the amount of control the user had on the effected sound. The Oakley DCR320 has no such buttons but instead allows full control over the most important parameters.

The DCR320 now provides controls for the wet/dry mix, the amount of left right cross processing, the modulation speed and depth, the modulation waveform and phase, and the delay time.

Input and output level control pots are also provided to allow maximum flexibility in dealing with a variety of input signal levels. Both the stereo input and output connections are balanced but can be used with unbalanced connections if desired. A mono/stereo switch is available on the front panel to allow just a single input to be processed in stereo.

In the DCR320's classic mode the unit features the original tonal equalisation and companding noise reduction circuitry which keeps unwanted noise levels to an acceptable level while also adding a 'vintage' sound of its own.

In the DCR320's clean mode the unit now gives a sound more reminiscent of the chorus units used in classic analogue synthesisers. The clean mode has no companding circuitry, no noise shaping EQ, and the effected signal has a wider bandwidth. Intermodulation distortion is significantly less in this mode which means pad sounds remain clear. However, the downside, like the original chorus units the clean mode tries to emulate, is that the audio outputs have a low level of swishing white noise that comes from using BBD devices.

A four LED level meter helps you keep signal levels at optimum ensuring the best signal to noise ratio without clipping.

The first prototype unit built into a Bryant Broadcast 1U 250mm deep 19" rack enclosure with the Schaeffer front overlay panel attached. This one has an internal toroidal mains transformer.

Operating Instructions

INPUT LEVEL and LED signal meter

The input level control adjusts the signal running through the DCR320. It affects both the wet and dry signals. With the pot fully counter clockwise the signal is completely shut off.

The signal meter shows the audio signal level after the input level control. If the signal is too high then the BBDs will be overdriven and produce significant distortion. If the signal is too low then the inherent noise produced by the BBD circuits will become very noticeable. The aim then is to have as large a signal as possible without introducing too much distortion.

The input level should be set so that the yellow 'high' LED on the signal meter will pulse in time with the input signal. If the red 'peak' LED lights up the BBDs will be probably distorting. However, depending on the type of signals that are going through the DCR320, overdriving the BBDs for very short intervals may still produce acceptable results. Indeed, overdriving the BBDs in this way can produce some interesting artefacts. No harm will come to the unit if the unit is run continually in the red.

It should be noted that in 'Classic' mode the input signal is compressed prior to be being sent to the BBDs to reduce the likelihood of overdriving the BBDs and to improve overall signal to noise ratio. The signal meter is monitoring the signal prior to compression so it is quite possible that you will not hear any distortion when in classic mode even with the peak LED lighting up. However, in 'Clean' mode, if the peak LED is illuminating, even for a brief moment, the BBDs devices will be clipping the signal.

Stereo/Mono (Switch)

The DCR320 is a 'stereo in, stereo out' effects unit. It has two sets of input sockets and two sets of output sockets on the rear panel. When this switch is in 'Stereo' mode the audio signal on both inputs sockets pass to their respective audio channels for processing. When the switch is in 'Mono', the signal sent to the 'Right In' socket is ignored, and the signal sent to the 'Left In' socket is processed by both the left and right channels of the DCR320.

If only a mono signal is available, that is, you only have one cable, this can be connected to the 'Left In' socket. The single mono input signal will then be automatically applied to both the left and right inputs. In this case the Stereo/Mono switch will have no effect on the signal since the input signal is already mono.

Note: If a mono signal is connected to the 'Right In' input socket only and the Stereo/Mono switch is put into the Mono position then you will hear no audio from the output of the DCR320. This is because in Mono mode the DCR320 is only processing the signal sent to the 'Left In' input socket.

CLASSIC/CLEAN (Switch)

The DCR320 provides a choice of two distinct operational modes, 'classic' and 'clean'. The basic difference between the two is that the classic mode provides noise reduction while the clean mode does not. BBD devices generate a significant amount of noise and distortion so many vintage effect units that utilised BBDs would incorporate different types of noise reduction circuits. Although these circuits would reduce the level of hiss and whines heard, they would also introduce their own artefacts. Those artefacts would themselves often be part of the sound of the units in question.

The classic mode features an audio pathway that is similar to that of a well known studio rack effect with four preset modes. The signal is first put through a circuit that emphasises the high frequency components of the sound before being sent to a compressor with a fast attack and relatively quick release time. The compressor adds a significant sonic footprint to the sound of the unit particularly on those inputs that are both complex and sustaining, for example, synthesiser pads. While the high frequency pre-emphasis circuit can create significant distortion with input signals that are rich in harmonics.

In classic mode, the now compressed and high frequency boosted signal, is then low pass filtered to reduce aliasing, and sent to the BBD devices. The delayed signal is then low pass filtered again to remove high frequencies generated by the sampling process in the BBD. And then travels to an expander which attempts to reverse the compression effects applied at the input. The signal then has the high frequencies de-emphasised, again to reverse the effects of the pre-emphasis at the input. There is now some additional bass cut applied before being added to the original input to make up the final audio output.

In clean mode the signal pathway is somewhat simpler and more akin to what you would have found in a synthesiser chorus effect. There is no pre-emphasis or compression before the BBDs, and no expansion or de-emphasis after the BBDs. However, the low pass filtering is the same as the classic mode. The clean mode will not have any compression artefacts nor suffer from distortion if the signal is particularly bright. It will, however, suffer from a low level of hiss and exhibit some degree of modulation signal breakthrough.

DELAY TIME

This control manually adjusts the delay time of the BBD devices. At its maximum setting the delay time is highest. Delay time can be varied from 2.8ms to 10ms with this control alone.

Both left and right channels are affected equally by the 'Delay Time' control. However, it should be noted that the precise delay time in each channel is slightly different because of the analogue nature of the two internal VCOs that control that channel's BBDs.

LFO RATE

This controls the rate of the low frequency oscillator (LFO). The LFO's output can, along with the 'Delay Time' control, alter the delay time. An LFO produces an output signal that changes in voltage with a repeatable and periodic shape. The LFO's frequency varies from 0.03Hz (slow) to 50Hz (fast). Faster LFO speeds produce more obvious effects even if the amount of modulation has stayed the same. Because of this it is often necessary to decrease the 'LFO Depth' control when increasing 'LFO Rate'.

WAVE

The wave switch selects the waveform output of the LFO. Two choices are available; 'Triangle' and 'Sine'.

The triangle waveform rises at a constant rate, reaches an apex and then descends at the same rate as it went up. Then at its lowest point it rises again and the process repeats itself. It is called a triangle wave because when viewed on an oscilloscope the trace looks as if it were made from series of triangles. When the triangle wave modulates the delay time the effect is to produce a pitch shift of the audio signal that appears to move between two different frequencies.

The sine waveform runs at the same frequency as the triangle wave but moves up and down in a way that slows down towards its highest and lowest points. It is called a sine wave because when viewed on an oscilloscope it looks like the sine mathematical function. When the sine wave modulates the delay time the effect is to produce a pitch shift that is very natural sounding and has no sudden changes in frequency.

Most classic chorus effect pedals and rack units use triangle wave modulation shapes. However, if you want a more natural vibrato effect, ie. when the 'Effect Mix' control is set to 100% wet, then a sine wave is preferred.

PHASE

The phase switch determines the phase difference between the LFO signal modulating the left channel and the right channel.

When 'Sync' is selected both delay lines will be modulated by the same LFO signal. Thus both delay lines will move up and down together and be synchronised.

When 'Invert' is selected one delay line will be modulated by an inverted version of the LFO signal. Thus as one BBD's delay time is increasing, the other BBD's delay time is decreasing.

The classic stereo chorus units use the 'invert' function. This creates a rich stereo effect as the sound in each channel is different.

LFO DEPTH

This controls the depth of the LFO modulation. The deeper the modulation the greater the change in the delay times. Fast changes in delay times are heard as a change in pitch by the listener. Because of this the LFO frequency also appears to change the depth of the modulation. It is often necessary to alter the LFO Rate as the LFO Depth is changed.

BALANCE

The original device on which the DCR320 is modelled is very unique. Unlike most chorus effect units the original device mixes together the dry and delayed signal with a version of the delayed signal from the other stereo channel. This form of cross mixing is very special and makes it difficult to emulate on standard stereo units. In the original device the ratio between the various mixes, ie. dry, wet, cross coupled, are changed according to which of the four buttons have been pressed. In the DCR320 the mix is fully variable. The 'Balance' control adjusts the ratio of mixing between the delayed channel and the other channel's delayed signal.

Furthermore, in the DCR320 the 'classic' and 'clean' modes affect these signals too. In 'classic' mode the signals are processed as per the original device and have a particular set of EQ curves applied. In 'clean' mode no such EQ curves are applied and it is simply the delayed signals that are mixed by the 'Balance' control.

With balance control set fully counter clockwise the signal heard is the delayed signal of that channel. With balance control set fully clockwise the signal heard is the delayed signal of the other channel, ie. the cross coupled signal.

In 'clean' mode the phase of the cross coupled signal is out of phase with the channel signal. That is, the other channel's delayed signal at the clockwise end of the balance control is inverted.

In 'classic' mode the phase of the cross coupled signal is in phase with the channel signal. This means that the channel's delayed signal at the counter clockwise end of the balance control is inverted.

This means that the behaviour of the 'classic/clean' switch has a marked effect on the action of the balance control. While this may seem somewhat counter intuitive bear in mind that the original rack effect that the DCR320 emulates works in a different way to the standard chorus units from the same company. In each of the DCR320's two modes the action of the balance control is the most natural for that mode.

For basic stereo chorus the balance control is usually expected to be in the fully counter clockwise position, ie. no cross coupling. That said, you get interesting effects with the balance control set to its mid point.

EFFECT MIX

This adjusts the mix between the non delayed signal and the one coming from the delay line or lines. 'Dry' is the non delayed signal while 'Wet' is the delayed signal. The input and output level controls affect both dry and wet signals.

In 'clean' mode the 'dry' signal is identical to that of the unprocessed input signal. So with 'clean' mode engaged and 'Effect Mix' set to dry, the DCR320 does not appreciably affect the audio signal. In other words the effect engine within the DCR320 is bypassed. However, in 'classic' mode the 'dry' signal is not truly dry. This is because the DCR320 copies the circuitry of the unit it emulates and the dry signal has a partial bass boost applied. This bass boost is to counteract the bass loss in the 'classic' mode's effected signal so when the dry and wet signals are mixed back together the overall tonal balance is maintained. So to obtain a truly dry signal the DCR320 must be switched to 'clean' mode.

OUTPUT LEVEL

This adjusts the output level of the unit. The gain of the whole module is +20dB when being driven with an unbalanced audio signal and both input and output level controls set to their maximum.

POWER LED

This will light when power is applied to the unit. To be precise, it lights when the +15V power supply rail is up and running.

Input and Output Connections

The DCR320 has both balanced input and output connections. It is expected that the unit will be fitted with three pole TRS (tip-ring-sleeve) sockets. Both input and output are compatible with unbalanced signals and mono jack plugs can be used without detriment to the DCR320.

It should work well with signal levels direct from a mixer, line level synthesisers and modular synthesisers. The input impedance is too low to work direct from guitars unless they are fitted with internal pre-amplifiers.

The DCR320's stereo inputs can be used with mono sources. Simply connect the jack plug to the 'left in' input socket and the input signal will be applied automatically to both the left and right inputs. In this case the Stereo/Mono switch on the front panel will have no effect on the signal since the input signal is already mono.

The maximum input signal level without clipping the pre-amplifier stage is +/-12V. Signal levels higher than +/-30V have the potential to damage the unit. Input impedance is 44K. Output impedance is 220R.

Calibration

There are six trimmers on the DCR320. It will be useful to have access to an oscilloscope and a frequency counter for the complete calibration routine.

All voltages should be measured with respect to a suitable 0V point. That is the black lead of your scope should be connected to 0V. 0V is most easily found at the anode of D3 on the DCR320 main board which is the end that is nearest to the power header.

LFO This sets the maximum frequency of the LFO.

Set your scope's time base to 5ms per division and the scaling to 2V per division. Set the LFO Rate control to maximum.

Monitor the voltage at pin 1 of U10. Adjust LFO until you get a 50Hz triangle wave. Any value between 48Hz and 52Hz will do.

TUNE This sets the minimum frequency of the two VCOs that control the delay time. The idea is to have the minimum frequency of the slowest VCO to be around 50kHz.

Set the DCR320 so that the LFO depth pot is at its minimum value, and the Delay Time pot is at its maximum value.

Measure the frequency of the signal at pin 2 U8. Adjust TUNE so that the frequency of the square wave is approximately 50kHz. Any value between 49kHz and 51kHz will do.

Now measure the frequency of the signal at pin 2 U9. If the frequency is between 49kHz and 51kHz then TUNE can be left alone. If the frequency is below 49kHz then adjust TUNE so that the frequency of the square wave is approximately 50kHz.

NULL1 & **NULL2** These adjust the amount of high frequency clock breakthrough into the audio output of each BBD. You want to set these so that the smallest amount of clock is getting through to the audio output.

Set your scope's time base to 5uS per division and the scaling to 100mV per division. Turn the input level down on the DCR320 so that no signal is passing through the delay lines. We need to monitor the voltage at the input to the first low pass filter for each of the four BBD circuits. These are pin 1 of U23 for NULL1, and pin 7 of U23 for NULL2.

Adjust the relevant NULL trimmer so that the peak to peak amplitude of the waveform seen on the oscilloscope trace is minimised. Set incorrectly you will see a series of alternate spiky sawtooth like waveforms, one bigger than the other. Set correctly, each alternating sawtooth will be mostly of the same size. Do this for both BBD lines.

OFF1 & **OFF2** These adjust the bias point of the respective BBD line's input signal. If this voltage offset is set too high or too low then the output signal of that BBD will distort too easily. The ideal point will be when the signal running through the BBD is at its maximum without any degradation at the waveform peaks.

Input a 220Hz triangle wave into the DCR320's 'Left In' input socket. Set the Stereo/Mono switch to Mono, and set the Classic/Clean switch to Clean. Adjust the input level so that the red PEAK LED is just lit. Set the LFO Depth control to its minimum, and set the Delay Time control to its middle position.

Set your oscilloscope's time base to 1ms per division and scaling to 1V per division.

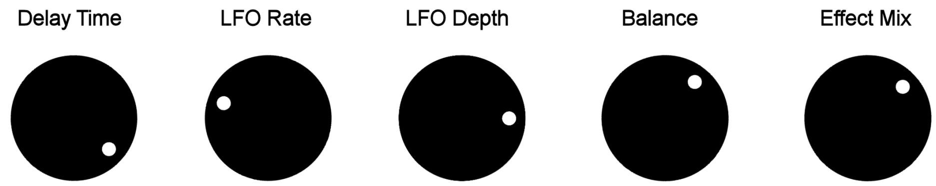
Adjust OFF1 so that the waveform seen at pin 1 of U23 is not clipping at either the top or bottom of the waveform.

Adjust OFF2 so that the waveform seen at pin 7 of U23 is not clipping at either the top or bottom of the waveform.

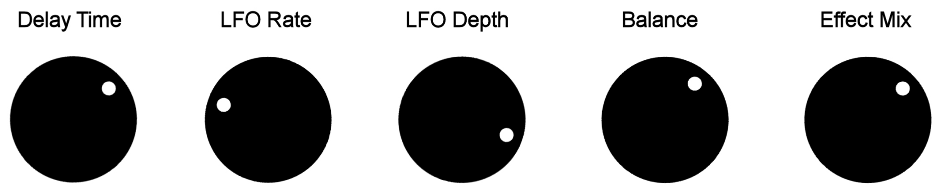
Example Settings

All four sounds have the mode set switch set to 'classic', the wave switch set to 'triangle', and phase switch set to 'invert'. Note that all settings are approximate as there will be some variation between units.

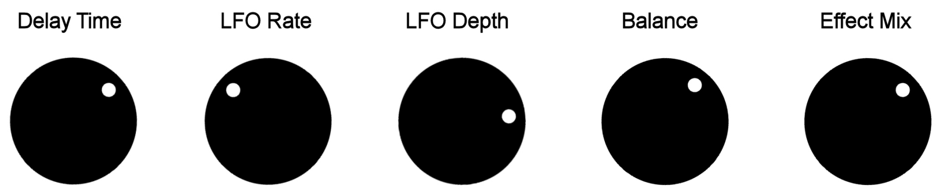
Classic Preset 1 – LFO rate 0.26Hz, delay time 8ms to 12ms.



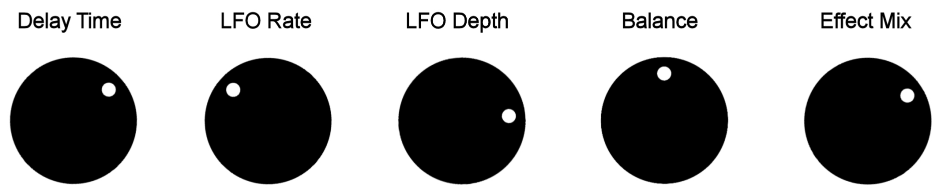
Classic preset 2 – LFO rate 0.26Hz, delay time 5ms to 10ms.



Classic preset 3 – LFO rate 0.56Hz, delay time 6ms to 9ms.



Classic Preset 4 – LFO rate 0.56Hz, delay time 6ms to 9ms.



Final Comments

I hope you enjoy using the Oakley Sound DCR320.

If you have any problems with the module, an excellent source of support is the Oakley Sound Forum at Muffwiggler.com.

If you have a comment about this user manual, or have a found a mistake in it, then please do let me know either via e-mail or the forum.

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