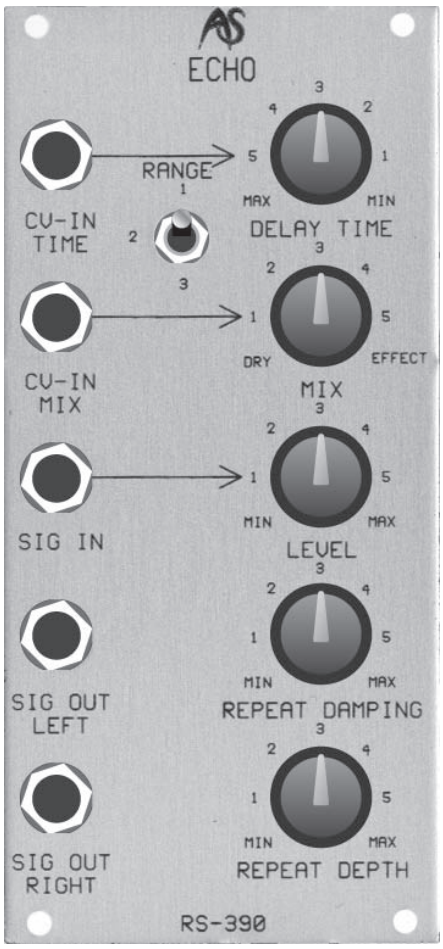
RS390

ECHO



INTRODUCTION

Echo was almost certainly the earliest 'effect' used on electronic sounds. It is also the easiest to produce: all you need is a tape recorder with a record head offset by a few centimetres from the playback head. If you then record a sound onto the tape, you can replay it as a single echo a few fractions of a second later, with the delay determined by the tape speed and the distance between the heads. Later innovations included machines with multiple heads that produced a series of echoes, and tape loop systems that - if you wished - extended the number of echoes to infinity, each sounding muddier and less like the original sound than the previous. If you listen to the electronic music of the 1950s, you'll find it awash with tape echo, sometimes used to excellent effect, more often not.

The problem with tape echo was that it was neither convenient nor cheap to produce. A lighter and more affordable alternative arrived in the 1970s with the development of the bucket brigade device, or BBD. Although totally analogue in nature, a BBD takes a series of samples of the incoming audio, and allows you to tap these at various stages as they pass down a series of discrete steps through the device. BBDs made cheap electronic delay lines a commercial reality and, although they never sounded as good as their tape-based counterparts, solid state "echo units" soon became a staple of electronic music.

Although the maximum delay times available from BBD echo units tended to be rather short - of the order a few hundred milliseconds - they proved to be ideal for a wide range of electronic effects such as chorus, flanging and phasing, whereupon their often metallic sound could prove to be a benefit rather than a hindrance. The Analogue Systems RS310 Reverb / Chorus is one such device and, unlike the stand-alone units developed in the '70s, this offers significant benefits such as voltage control of the delay time and voltage control of the mix between the unaffected and affected signals.

Of far higher fidelity, digital delay lines (DDLs) were also developed in the 1970s. Although rather clunky by today's standards, the earliest of these were unbelievably expensive, which is why they did not come to the attention of most musicians for another decade or so. To explain precisely how a digital delay line works would require a thorough treatise on sampling theory, and this manual is not the correct place for that. Nonetheless, it's not hard to grasp the basics…

A digital delay line is nothing more nor less than a specialised computer that samples an incoming signal and stores it in RAM. These samples are typically taken at a rate of 44,100 times per second, and stored with a resolution of 16- or 24- bits per sample. Once a sample is held, it can be read back at any time (or times) determined by the delay algorithm, until it is necessary to replace it with another incoming sample. The amount of time a sample can be stored is determined by the amount of RAM in the system.

If you can modulate the clock rate of a DDL and mix the affected signal with the original, you can create a much wider range of effects than just simple delays. As on their analogue counterparts, this is the method used to recreate sounds such as chorusing, flanging and phasing. However, few if any DDLs

(other than the RS390) have attempted to recreate the sound of a true tape delay, including the pitch shifts that are created when you change tape speed, and the others that occur as the recorded signal 'catches up' with the new speed. This is not surprising… it is a far from trivial task to do so. In contrast, the RS390 is a true emulation of traditional tape echo, with all of that technology's strengths and weaknesses. It allows you to generate sounds and extreme effects that are not achievable elsewhere.

IN USE

The RS390 is a 24-bit stereo digital delay line with a sample rate of 46.9kHz, a bandwidth of 22kHz, and a maximum delay time of approximately three seconds. If offers three delay ranges with CV control, as well as CV control of the dry/wet mix. The input and output converters (ADC and DAC) are 24-bit linear PCM although the RS390 uses 16-bit audio throughout.

DELAY TIME and the RANGE switch

If no CV is applied at the CV-IN TIME input, there are three delay ranges selected using the RANGE switch. These are:

1:	1.75ms	-	100ms
2:	8.7ms	-	500ms
3:	52ms	-	3,000ms

For each range, the minimum delay time (MIN) is to be found at the clockwise extreme of the control, and the maximum (MAX) is at the anticlockwise extreme.

The first repeat will appear in the LEFT channel after 1/2 the DELAY TIME has elapsed, to be followed in the RIGHT channel a further 1/2 DELAY later. Thereafter, the signal will ping-pong between the channels, with each repeat occurring 1/2 DELAY after the previous. This means that, with the exception of the first repeat in the left channel, the delay in a given channel is equal to the DELAY TIME.

MIX

With no CV applied at the CV-IN MIX input, the MIX control determines the relative volumes of the original (DRY) signal and the affected (EFFECT) signal. The amplitudes are equal when the control is in the 12 o'clock position.

LEVEL

This adjusts the amplitude of the internal digital signal generated from the audio presented at the SIG IN input. It acts in the digital domain, and offers gains from -∞dB to approximately +3dB. For optimal audio performance this should be used as close to unity as possible.

REPEAT DAMPING

This is a low-pass filter within the feedback loop that generates the repeats. Reducing the high-frequency content of each successive repeat emulates the response that you obtain from a tape echo unit.

- REPEAT DAMPING = MAX

  Maximum filtering is applied. There will be a single repeat in the right channel and a single repeat in the left channel.

- REPEAT DAMPING = MIN

  No filtering is applied.

REPEAT DEPTH

This controls the gain of the feedback loop that generates the repeats.

- REPEAT DEPTH = MAX

  All of the delayed signal is fed back to the input.

- REPEAT DEPTH = MIN

  No signal is fed back to the input, and there are no repeats.

Note 1: The number of repeats is dependent upon both the REPEAT DAMPING and REPEAT DEPTH controls. Reducing the damping or increasing the repeat depth will increase the number of repeats.

Note 2: If you set both the REPEAT DAMPING and REPEAT DEPTH to MAX and apply an impulse to the signal input you will hear the original signal, then a single repeat from the right output, then one from the left, followed by a pair of dull thumps which are all that is left of the signal after it has passed through the low-pass filter.

Note 3: You can set the controls so that the repeats continue ad infinitum. This can lead to interesting sonic effects, but it can also cause distortion within the RS390, as well as high output levels that may cause distortion further down the audio path.

Inputs and Outputs

The RS390 offers an audio signal input, dual Left/Right audio signal outputs, and two CV inputs.

CV-IN TIME

Accepts CVs in the range ±5V, and allows you to sweep the selected RANGE across its entire range of values. The incoming voltage is converted to a digital control signal with 10-bit resolution.

CV-IN MIX

Accepts CVs in the range ±5V, and allows you to sweep the MIX across its whole range. The incoming voltage is converted to a digital control signal with 10-bit resolution.

SIG IN

Accepts audio signals in the range ±10V.

SIG OUT LEFT and SIG OUT RIGHT

Output audio signals in the range ±10V