

Science In The Studio: FlexDelay VST plugin

Disclaimer

FlexDelay is free audio processing software which is provided without warranty and no responsibility is accepted for loss or damage arising from its use. By using this software you are agreeing to these terms of use.

What is this document?

This short manual describes how to install and use the FlexDelay plugin. It also explains what the plugin does to audio.

What is FlexDelay?

FlexDelay is a digital delay tool that can be used in any VST plugin compatible program running on a Windows PC (such as *Cubase*, *Reaper* or *Wavelab*)¹.

There are loads of delays out there, why do I need FlexDelay?

This is a free delay which, as its name suggests, is designed to offer a lot of flexibility in setting its parameters. If you already have a delay which offers control of delay time, feedback, modulation rate and modulation frequency with a wide range of settings for these parameters, then you probably don't need FlexDelay. Otherwise, read on.....

A delay is just a delay isn't it?

Yes and no. At its simplest, and most widely recognisable, a delay is a device for producing an echo effect: audio played through it is repeated later in time (again and again, if there is feedback). But the process of delaying a signal and adding it to itself is at the heart of many other processors, such as equalisers, chorus units, flangers and reverberators. Some, but not all, delays are flexible enough that they can be configured to produce chorus, flanging and comb filtering. FlexDelay is one of these delays. It also offers extreme parameter settings that can produce Doppler/pitch shifting effects.

What is the connection with 'Science In The Studio'?

Time and frequency are inextricably linked. The time taken for one cycle of a waveform is directly related to the frequency of that waveform. As the time taken (the 'period') *increases* so the frequency (the number of times the waveform repeats in one second) *decreases*. In the same way, an audio processor that delays a signal in time, can also have an effect on the frequency content of that signal. This plugin allows this relationship to be explored, either in the classroom/audio laboratory or the studio. A really good grasp of this relationship is something that recording engineers need if they are going to be able to explore the full potential of what is known as 'time domain processing'. The idea of *Science in The Studio* plugins is to provide tools that can be used for both learning and as a studio effect.

¹ Any 32 bit Windows application that can support VST 2.3 format plugins, to be precise.

What happens when we delay a signal?

This largely depends on two things. The first is whether the sound is continuous, i.e. 'stationary', (such as a sustained organ sound playing the same note) or rapidly changing, i.e. 'transient', (such as a short snare drum hit). The second is how long the delay is from when we hear the original sound and when we hear its repeat. For a transient sound with a relatively long delay (longer than 50 milliseconds or so) then the delay is heard as a distinct repeat. When the delay is shorter than this, or the sound is continuous and changes only very slowly over time, then we hear changes in the spectrum (or frequency content of the signal). This is because when a sound is mixed with itself under these conditions the mixture is heard as one fused sound, rather than as two separate sounds occurring together. The frequency content of this fused sound is affected by what is known as constructive and destructive interference: at some frequencies the waveforms of the two sound components are moving together (i.e. are 'in phase') and so the combined result is a relatively high amplitude at these frequencies. At other frequencies the waveforms of the two components are in opposition to each other (i.e. are 'out of phase') and they cancel each other out to a certain extent, giving a combined result that is quite low in amplitude. When the delay is constant then this effect is known as 'comb filtering' since peaks appear in the spectrum of the resultant sound and these peaks are equally spaced along the frequency axis (resembling the teeth of a comb).

If the delay time is varied then the position of these teeth is constantly moving up and down the frequency axis, creating a sense of movement in the spectrum and a regularly changing sound. With certain settings a flanging effect can be produced – this is often described as a 'swooshing' sound. With other settings a chorus/ensemble effect can be produced and with quite extreme settings where the delay time is changing very rapidly then a Doppler effect is produced where the changing delay causes the waveform to be squashed together or stretch apart causing variations in the pitch! More information about time domain effects can be found in books such as Curtis Roads' *Computer Music Tutorial* or David Miles Huber's *Modern Recording Techniques*

How do I install and access the plugin?

All you need to do is place the FlexDelay.dll file in the VST plugins folder of your host program (e.g. *Cubase*). If you're not sure where this folder is, check the documentation for your host software. Once you have installed the file, it should be accessible within the channel strip of the mixer in the host. Again, if you're not sure how to load effects in your host software have a look at the documentation that came with it. The user interface for the plugin when used in the host program *Cubase LE* is shown overleaf.

Using FlexDelay

This effect can be used as an insert effect or with an auxiliary bus. When using a time-domain processor, such as a delay, with a mixer in a program such as *Cubase* then it is standard practice to use it with an auxiliary bus. This allows once instance of the plugin

to be used on many different tracks of audio via the same auxiliary bus. Using it as an insert would require a separate instance of the effect in every track/channel.



The parameters

Wet: The proportion, as a percentage of the input signal that is fed into the delay. If the delay is used with an auxiliary bus then setting this to 0% will result in no output from the unit. If it is used as an insert effect then setting this to 0% will result in a completely un-effected signal being passed through the device.

L Delay: The delay time for the left channel (in milliseconds).

R Delay: The delay time for the right channel. The delay times for both channels can be set completely independently of each other.

Multiplier: This acts on the delay time. When the multiplier is set to 1.0 (its maximum setting) the range of delay times available is 0 to 1000 ms (0 to 1 second). When set to a value less than 1.0 then the ranges of delay times are scaled by this amount. This means that very fine control of the delay is possible at low delay (just set the Multiplier to a low value).

L F'back: This is the proportion of the output from the left channel delay, as a percentage, that is fed back into its input. A low setting will cause repeats of the signal to die away quickly, a setting of 100% will cause input sounds to be sustained indefinitely.

R F'back: As for L F'back, but controls feedback for the right channel.

L LFO: This controls the frequency of the low frequency oscillator which can be used to automatically vary the left delay time. The waveform of this oscillator is a simple sine wave and so it varies the delay time smoothly at whatever frequency this control is set to.

R LFO: As for L LFO, but controls the LFO for the right channel. This is completely independent from the LFO for the left channel and so can have completely different settings. This allows the creation of quite striking stereo effects!

L LFO Depth: This controls the extent to which the LFO varies the delay for the left channel. If set to 100% then the LFO varies the delay time between 0 ms and the delay setting shown for L Delay. If set to 50% then the LFO varies the delay time between half of the L Delay value and the full L Delay value. If set to 0% then the LFO has no effect on the delay time.

R LFO Depth: The same as L LFO depth but for the right channel. Again, this control can be set completely independently to the left LFO.

Some things to try

A typical flanger setting:

Delay time: around 10 ms

Feedback: the higher, the more pronounced and metallic the effect.

LFO frequency: you decide, this controls the rate at which the teeth of the comb move up and down.

Question: how does the delay time affect the spacing of the teeth of the comb filter?

A typical chorus setting:

Delay time: around 30 ms

Feedback: none

LFO frequency: you decide, lower settings will produce less of a detuned effect.

Question: how is the delay simulating the sound of 'chorus' or ensemble?

Settings for extreme Doppler shifting:

Try to discover these for yourself. Have fun!

Who made FlexDelay?

This plugin was produced by Jez Wells at the Audio Lab in the Department of Electronics at the University of York. It is designed for creative use but also so that the 'science in the studio' that recording engineers use in their everyday work can be investigated, explored and, ultimately, understood in the same in-depth way that engineers themselves do. Jez is currently supported by a Public Engagement Scholarship from the Royal Academy of Engineering. If you have any questions, comments or bug reports please email them (along with details of how you are using the plugin and the host application) to jez@york.ac.uk – thanks!

VST PlugIn Technology by Steinberg.

