hudak is a "multitrack" delay line system for very long delays. it was originally built for the composer john hudak (hence the name). it was originally intended for recording individual sounds that will replay at long, varying intervals. it can ofcourse be used in other ways, but is best suited to long delay times. for working with short delay times (10 seconds down to 10 milliseconds) it's better to use "decaydance" (available on my web site)

these instructions assume no knowledge of Max/MSP. so you may need to skip some bits if you know these programmes.

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the 6 delay lines are arranged in two groups of three, with one group fed from the left input of the computer, and the other from the right input.

this means that you can feed seperate sound sources into each line. alternatively, by clicking on the [left->both] box, you can feed the left input to both lines.
to change the input source, double click on the [dac~] box (lower right). you can select a different input source in the drop-down menu. (if you start the program and don't seem to be getting any audio input, check here first. switch input to Sound In or Internal Mic - but be careful of feedback if you select the mic!)

midi input is not implemented at the moment.

the maximum length of the delay lines, and the internal buffers (in [rec2bfr] and [grain]), is dependant on the amount of ram allocated to the program. basically - give it all you can (leaving around 5meg for the system to expand in).

the lengths that you will be able to set depend on how much memory you have given the patch; each line can be up to 2 minutes long, and the buffer in [rec2bfr] may be up to any length.
every time that a [max delay] value is changed, or the [rec2bfr] buffer is resized, the amount of available memory remaining is shown in the [freemem] number box. interpreting this is something that you'll have to establish by trial and error.
keep the Max window open (<command>-m) while altering max delay times and keep an eye on the available memory number in the top right of that window. if you set the delay/buffer times to more than the

available memory, you'll get a warning in the Max window. back off the delay time that you were just setting and increase it one 'notch' at a time until you get another memory warning, then back the delay off by 1 second.

if you need to get the Mac menu bar back, click on "hudak" in the blue background. click again to remove the menu bar.

in any window, click on [front] to bring that window forward. typing numbers directly into boxes is only possible when a window is at the front. apart from that, all the windows are active all the time - you can scroll number boxes, change fader positions, and click on check boxes at any time, irrespective of which window is at the front.

note that if a number box is active (it's in the front window and has been clicked on), you can use the arrow keys (up & down) to change values. once you have done this - click anywhere else in that window to deselect the box, otherwise you may find values changing unexpectedly!

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to start [hudak], click on the large on/off checkbox. this turns audio processing on and off. some activites (those not involving audio) will continue even with audio off (eg autopan).

next fade up the input faders to about 75%.

if you want to hear your audio input before you start to listen to the delay lines, change the [thru] level to 1.0 (or as required). you'll probably want to leave this at this level, so that you can hear a sound as you play it (otherwise you won't hear it until it reaches the delay output). this control defaults to 0. to avoid sudden feedback when you turn audio on.

click on the small box next to [display]. this opens up a window which gives a rough graphical indication of the delay line and the taps on it. the upper bar in each pair represents the current maximum delay time for each delay, and the lower bar represents the current actual delay time.

main window

considering each vertical line of controls as a "track" and starting from the row of green buttons....

the green buttons are individual input mutes for each track; by default each track is on, ie receiving audio from the inputs. click once on a button to mute input to that track.

[clear] immediately clears the delay line for that track.

[max delay] sets the maximum delay time for that track, up to a maximum of 120 seconds. if you're using the randomiser, it will not generate values for a track greater than the amount set here.

[delay] is the actual current delay time on that track.
the reason for having these 2 different settings is so that the delay time can be randomly varied without altering the length of the delay line itself, which can result in audio artefacts.

back to the top of the tracks…

the top 3 controls belong to a delay time randomiser.
the checkbox turns the randomiser for that chanel on/off. note that the delay time will only change when audio processing is turned on.
the 2nd box down is for display only, and shows the current delay time when the randomiser is active. when you turn a randomiser off, the delay time is reset to the value in the [delay] box further down (the value in the [dly time] box isn't updated however).
[+/-] determines the amount that the delay time is varied by.

in practice - you first set the maximum delay time for a track, which should be the most that you will be likely to want (if you want to vary delay time 5 secs above and below 50 seconds, then you will need to set the max time to (50+5)= 55 seconds).
then set the current [delay] time for the track.
if you want the time to vary 10 seconds above and below this time, set the [+/-] value to 10.

set the [time between changes] box to whatever value you want. (longer times are probably better - start with the default of 45 seconds and work out from there).

click on [on/off] and the delay time will start changing. you can see

this easily in the [display] window.

back to the middle.......

[feedback] determines the amount of signal fed back from the end of the delay line to the beginning, thus giving extra repeats to short sounds or sustaining long sounds.

if feedback is set to [0.], any sound will play once only through the delay line. as you increase this value (to a maximum of 1.0) more of the sound will be fed back to the beginning of the line, and will be re-repeated. values around 0.8 mean that sounds will repeat for a long time (many minutes!).

[mute/vol] - clicking the mute checkboxes mutes the output of that delay line. audio is still being fed into the track (unless the green mute button is pressed). the little fader next to the checkbox is the output level for that track. if you have set the level for a track, and you don't want to change it, using the mute button is a quick way of closing that track without losing your level settings. the [fade time] box at the right of this line sets the time taken for the fade down/up when the mute button is clicked. it's in milliseconds, so a value of 2000 is equal to 2 seconds.

bottom 3 controls.....

these are the pan controls. use them to set the stereo position of each delay. if you click on the [autopan] buttons, the pan position will change by itself. the rate at which this happen is set by the number box at the bottom. note that alternate autopans counter-rotate.

the [All] button is a master on/off, and the box below it is a master speed control.

to store a particular group of settings, use the small memory box underneath the [hudak] title. shift-click on a slot to store settings, single click on a slot to restore settings.
IMPORTANT - settings are not stored when you quit the programme. you will need to click on [write] to save a settings file, and [open] to load in a previously saved set.

the audio output box

record to disc

underneath the [dac~] are a set of controls for recording the output of the looper straight to hard disc.

first click on the upper [open] box. a standard Mac dialogue box opens allowing you to create and name your audio file. do this every time before you start to record - if you don't change the file name, previously recorded audio may be recorded over.
click on the [rec] box to start recording, and again to stop.

to replay an audio file, click on the next [open] box (below rec2disc), and select the file you want to play. when you click on [ok] you get another Open file dialogue - the same audio file should be highlighted; just click on [ok] again.
the number box allows you to enter a start play time (in case it's a long file and you don't want to start at the beginning), and the checkbox at the end turns play on & off.
the 2 boxes underneath are for display only, and show the length of the audio file that you have loaded.

VST plugins.

you can load a vst plugin into the output path.

to load a plugin, click on [plug].
to open the plugin's editing window, click on [open].
the wet/dry fader crossfades between the clean and the treated signal.

if nothing seems to be happening after you have loaded a plugin - turn audio off and back on again. (this sometimes happens the first time (only) after you load a plugin whilst audio is already running).

built-in effects.

rec2bfr is a module for recording audio to an audio buffer that can then be loop played. the advantage of looping sound in this way is that the replay speed can be altered, or reversed, slowing/speeding or reversing the audio in the buffer. (values between 1.0 and 0. slow the audio, negative values reverse it).

first select the source using the dropdown menu in the top left. you can select audio left, right, delay lines 1-3 or 4-6, or grain output. perhaps you have a nice loop running in one of the set of delay lines and you'd like to loop it indefinately, and build up a new delay pattern. select the delay group that you want to record.
check the input level on the meter and adjust the input level as required.
set the buffer size (how long you want to record the audio). (note that this uses ram, so the size of buffer is, like the delay lines, dependant on available ram).
fade up the output fader (right side of module).
when you're ready, click on the [record] checkbox. the module will go into record & play (though you won't hear anything until the end of the record time) and the clock will start counting down (this is to show you the remaining recording time; so that, for example, if you're recording live into the buffer, you know how much time is left to record.)
you can turn play off/on with the [loopplay] box.

grain.

click on the grain checkbox to open the module.

grain is a cutdown version of a patch called Granular Synthesis, written by nobuyasu sakodna.

the output of grain is fed directly to the output faders.

select the input using the menu box in the top right. if there is audio on the selected input, you should see it on the left hand meter. you can adjust the input level using the number box [input].
click on the [rec] box - notice that the on/off and loop boxes in the sequencing section also come on.

after a short while you should hear the audio output of grain. you can change the pitch of the sound output by changing the value in the [pitch] number box.
you can change the playback speed in the box of the same name in the sequencing section.
if you turn playback to off, while still leaving loop checked, then the bit of audio at that playback position will continue to play. change the [grain dur] and notice that the audio loop gets longer/shorter. if you leave playback off while record is still on, then the sound being looped

will change every so often as the bit of audio at that point in the record buffer gets overwritten.
slower playback speeds sound better with longer grain durations.
the output of the 4 grain players are spread across the stereo image. if you click on [sep out], the 4 grains are mixed together resulting in a smoother sound. this sometimes sounds better with shorter grain durations (it depends on what effect you want).
if playback is off, you can change the play position by scrolling the position number box directly.
you can also use the [LFO] to control the play position. open the module by click on the [LFO] button. connect the LFO output to the play position by clicking on the [off] button.

LFO.

there are 2 wave generators in this module: a saw/triangle and a random generator.
use the [source] dropdown to select wave, random, or both.

by dragging in the upper or lower halves of [output range] (in the random section it's the lower of the 2 long boxes) you can set the upper and lower limits of the LFO range.
the [output display] boxes show you what's going on.
[start] starts the LFO.

the [period] number box sets the period of the triangle wave. the dropdown menu to the right selects the waveform.
the [lo] and [hi] number boxes set the range of the random number generator.

grain buffer.

by double clicking on the [buffer~] you open a window which shows you the audio in the buffer. note that this does not update in realtime; it show the audio at the moment the window was opened. to close the window click on the macintosh close box in the top left of the window.

to save the audio currently in the buffer, first turn off (grain) record.
the click on [wr]. a standard mac dialogue box will open, allowing you to save the buffer to disc.
click on [N] to normalise the contents of the buffer. note that you can do this whilst in record and/or play. if grain is playing, and the audio in the buffer is low level, the audio output will get much louder

immediately you click on [N]. you might want to fade the output of grain down first!
click on [t] to put a little fade on the start and end of the audio in the buffer.
[clear] immediately clears the contents of the buffer.