

Time Module (option)

The Time Module is an optional set of DSP processing tools included within the software. The three functions are, **Time Stretching/Compression** (no change of pitch), **Pitch Shifting** (no change of time) and **Sample Rate Conversion**.

The tools can be dragged into a toolbar using 'Customise toolpage' from the **Settings menu** or by double-clicking in a free part of the toolbar.



Time Stretch/Compress Process Tool

The **Time Stretch/Compress Process tool** allows individual or stereo linked parts to be changed in length without any pitch variation. This could be used, for instance, to reduce a TV or radio commercial soundtrack from 31 seconds to 29 seconds. New take(s) are always generated to disk for the selected part or stereo linked parts and processing typically takes 3 times longer than the length of the stereo part being processed. This amount of processing time is normal for a process such as time stretching, as large amounts of 'number-crunching' are required to achieve acceptable quality.

Typically, the algorithm is able to produce good quality results with a +/-20% change in time for most kinds of complex material (e.g. full frequency/dynamic range mixed music with vocals or voice-over), but this can be more for audio which has simpler dynamics or harmonic content.



If you require a severe change, it may be better to process the part twice with more moderate settings in order to reach the desired result.

Remember though, that it is impossible to produce a perfect algorithm. Due to the nature of the task, there will always be a difference between the original and time-altered versions. The Soundscape algorithm is, however, able to produce extremely high quality results which equal or surpass the results obtained from many 'high-end' (*i.e.* high cost) systems.

The Soundscape algorithm produces an unchanged left/right phase response when processing stereo linked parts. Other systems may produce a varying delay between left and right channels, which means that centre panned tracks (e.g. vocals) move around in the stereo image, obviously a very undesirable artefact.

Time accuracy is another important criteria for ensuring that, especially when working with music tracks, the rhythmic 'feel' of the music is not disturbed. Here also the Soundscape algorithm has extremely good performance when compared with that of other systems.

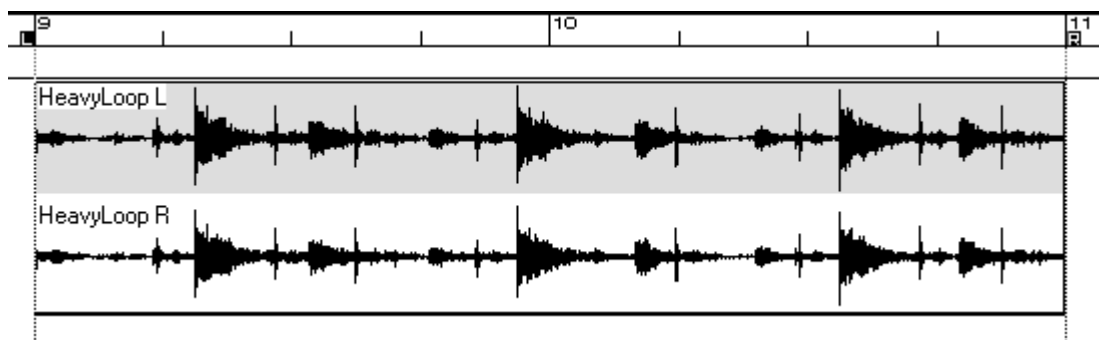
The time variation may be entered as a percentage of the original length of the part (limits are 67% to 150%), or as a new absolute time (SMPTE/Bars & beats) for the processed part(s), or as new tempo value (assuming that you

Soundscape

have determined the original tempo). The absolute time allows a 'stretch to fit' if locators are placed around a 'gap' and the readout of time between L & R locators is entered.

Entering a value in one field will not update other values until you click on one of the other fields.

The waveform below is a drum loop with a tempo of 87.22 bpm.



To stretch the drum loop to 85 bpm, click on the part(s) with the time stretch/compress tool. The following dialogue box will be shown:-

Time Stretch/Compress ✕

Part: HeavyLoop R

Stretch/Compress Ratio (%): 102.61

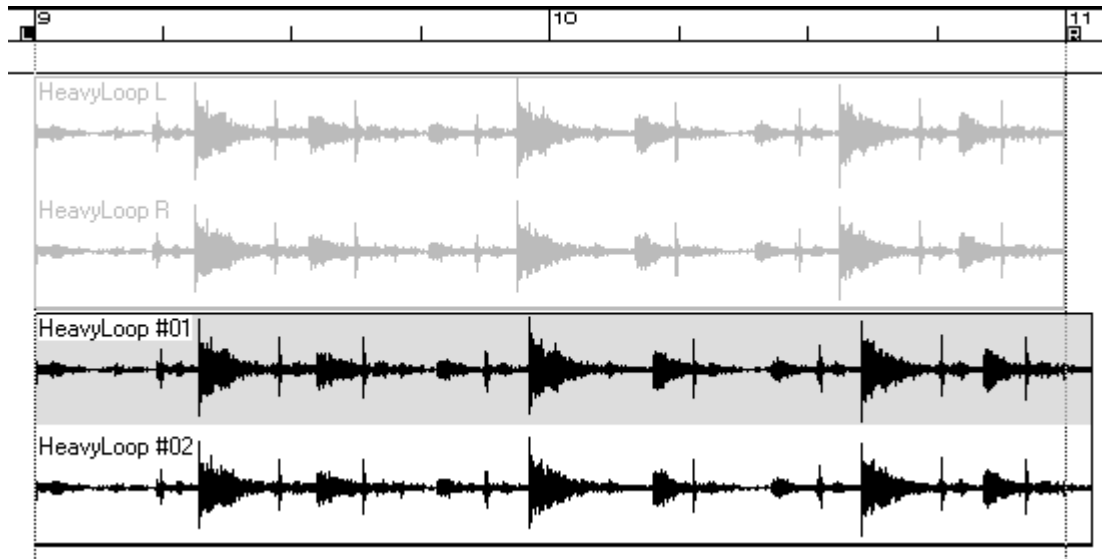
Original Part length:		Stretched Part length:	
00:00:05:24.1312	SMPTE	00:00:06:03.1155	SMPTE
2. 0.135	B/C/T	2. 0.179	B/C/T

Original Tempo: 87.22 **Converted Tempo:** 85.00

After clicking on **OK**, a new part(s) will be generated in the same way as all other disk based processes.

Note: It is important to remember that "stretched" parts will be longer than the original so you need to ensure that the parts to be processed have space after them.

The following illustration shows the time stretched parts under the original. You can see that the new parts are slightly longer and the drum groove is slower.



If a small change to the length of a part is required and a small change in pitch is acceptable, then a sample rate conversion will always produce superior results to time stretching/compression.

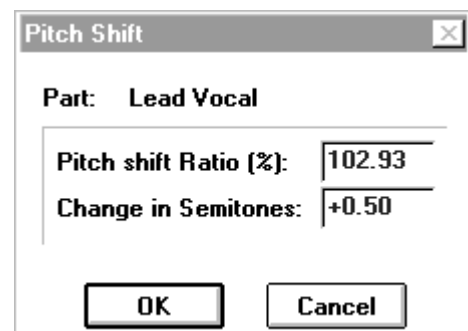


Pitch Shift Process Tool

The **Pitch Shift** tool performs in a similar way to the **Time Stretch/Compress** tool, except that it can vary the pitch by up to +/-7 semitones (approx.), without changing the length of the part(s).

Pitch values can be entered in semitones and cents (1/100s of a semitone) or as a percentage of the original (67% to 150%).

Pitch shifting is a two-stage process, which means that the blue bar in the processing box will run across twice. If you experience a delay (from 20 seconds up to a couple of minutes) after the first pass, before the second pass starts, then it will be because your PC does not have a maths co-processor (487) installed (*i.e.* for 486SX PCs).



Note: *The maths co-processor provides a table of floating point values which are used by the DSP during the calculations. If no maths co-processor is available, then the main 486 chip has to calculate these values, and it is not particularly fast at this kind of mathematical task.*

Our advice is to install a maths co-processor chip, as it will also allow EQ curves to be updated much more rapidly.

Note: Although you can enter a pitch shift of 12 semitones, the maximum shift is 7. It is possible to shift a sound an octave by shifting 6 semitones and then shifting the new part by 6 semitones.



Sample Rate Convert Process Tool

The **Sample Rate Conversion** tool processes an individual part or stereo linked parts and generates a take which is 're-sampled' at a new sample rate. This allows takes recorded at one sample rate (e.g. 48kHz) to be used in an arrangement with other takes recorded at a different rate (e.g. 44.1kHz).

The algorithm produces extremely high quality results by using 128x oversampling and sine interpolation. Unlike real-time sample rate converters, there is no requirement to restrict the calculation accuracy to enable real-time processing. There is **virtually no detectable change in sound quality** after processing, except for a change in bandwidth when converting to 22.05kHz or 32kHz. This means that the algorithm is ideal for converting 48kHz tracks from DAT to 44.1kHz for CD mastering, also for converting 48kHz Takes to 96kHz.

Samplerate Conversion	
Part: Final Mix	
Conversion Ratio (%):	91.88
Original Part length:	Converted Part length:
00:00:17:13.1452 SMPTE	00:00:16:03.0298 SMPTE
6. 1. 99 B/C/T	5. 3. 85 B/C/T
Original Samplerate: 48000	Conv. Samplerate: 44100
OK	Cancel

The new sample rate may be entered as a percentage change from the original (from 50% to 200%) or as a new sample rate. If one of the 'standard' rates (selected in the status bar) is entered, the new take will be shown with this sample rate in the **SFile Manager**. If a non-standard rate is entered, the SFile Manager will show the sample rate used for the current arrangement. If the original take has an 'unknown' sample rate, then the current sample rate for the arrangement is shown as the original rate in the dialogue box.

If you wish to convert 44.1kHz takes to 11kHz for multi-media applications you should do it in two passes - first convert to 22.05kHz and then to 11kHz. The audio quality will not be adversely affected by processing a number of times.

Note: A maths co-processor is also desirable for sample rate conversion as a floating point table is required (see Pitch Shifting).