DIGITALmusic online tutorials on conputer music
A project by UNESCO and University of Plymouth
Author: Eduardo Reck Miranda

Editing

Editing Samples

After a sound is digitally recorded (i.e., sampled) one can edit it in a number of ways. Editing procedures can be classified into two large groups: manipulation editing and function editing. This lesson introduces the most common of these procedures. An editing program is used to edit samples that are stored in a sampler or in a computer. Essentially, most keyboard synthesisers on the market can be considered as samplers because they often have samples of various instruments in their memory. Once the sample is recorded, however, it may need to be modified for a specific use or mixed with other samples. Basic modification of a sound sample could be useful for removing the presence of unwanted noise or selecting only a portion of the sound sample due to an interest in hearing only a part of it; for example, to alter the starting or finishing point of the sample.

Manipulation editing

Manipulation editing does not process the digital representation of a sample directly. Examples of manipulation editing include cut, copy and paste, increase playback speed, decrease playback speed, and looping. Manipulation editing includes the ability to cut and/or copy a portion of a sampled sound. When a portion of a sound is cut or copied, it is placed in a buffer to be re-used later, similar to what happens with text in a word processor program. Material may be shifted or copied from one portion of a sample to another; this function is called splicing. Splicing simply places the start of the material being added at an intersection point, butting up against the end of the previous material.

manipulation editing

Original sample

Sustain loop: Allow sample to continue while key is held
Release loop: Allow sample to continue after key is release

Cut

Join two samples end-to-end to create a new sample
Butt: Move abruptly from one to the other
Crossfade: move smooty from one to the other through a transitional area
containing elements of both samples

Copy and Splice

Remove unwanted sound or silence before desired sound

Increase Speed

Remove unwanted sound or silence after desired sound

Decrease Speed

Shape amplitude, add release

Reverse

Reverse direction of some portion of a sample:

- Example 1
- Example 2
- Example 3

Another typical sound editing procedure is looping. Looping becomes important if a sample needs to last longer than the original event. There are basically two types of loops: forward loops and reverse loops.

Function Editing

In contrast to manipulation editing, function editing processes the digital representation of a sampled sound. As its name suggests, it involves the use of functions (i.e., computer programs) to manipulate the samples of a sound directly. Examples of function editing include gain change, normalisation of amplitude and filtering.

Gain Change and Normalize

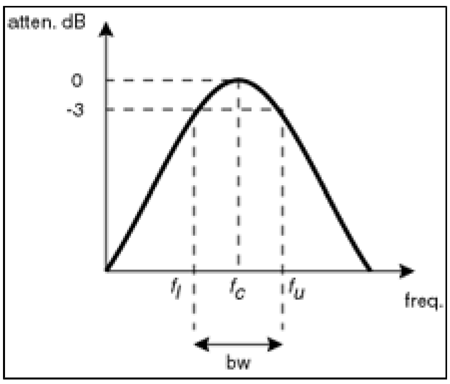
Gain Change: Adjust the amplitude of some portion of a sample
Normalize: Set the amplitude of the loudest portion to 100% and scale the rest accordingly

Filtering

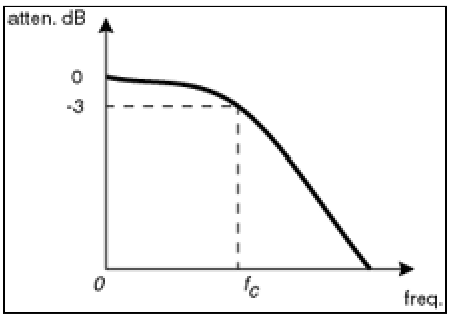
In general, a filter is any device that performs some sort of transformation on the spectrum of a signal. For simplicity, however, in this tutorial we refer only to filters that cut off or favour the resonance of specific components of the spectrum. Generally speaking filtering affects the physical nature of the manipulated sound. In this case, there are four types of filters, namely: low-pass (LPF), high-pass (HPF), band-pass (BPF, Peak) and band-reject (BRF, Notch). The BPF, also known as the resonator, rejects both high and low frequencies with a passband in between. Two parameters are used to specify the characteristics of a BPF: passband centre frequency (represented as f_c) and resonance bandwidth (represented as *bw*). The bw parameter comprises the difference between the upper (represented as f_u) and lower (represented as f_l) cut-off frequencies. The BRF amplitude response is the inverse of a BPF. It attenuates a single band of frequencies and discounts all others. Like a BPF, it is characterised by a central frequency and a bandwidth; but another important parameter is the amount of attenuation in the centre of the stopband. Equalisers are normally built by using an array of parallel BPFs. An LPF permits frequencies below the point called the cut-off frequency to pass with little change. However, it reduces the amplitude of spectral components above the cut-off frequency. Conversely, an HPF has a passband above the cut-off frequency where signals are passed and a stopband below the cut-off frequency, where the signals are attenuated.

Peak - Band-pass Boost selected band	f_c-bandwidth-amount
Notch - Band-reject Reduce selected band	f_c-bandwidth-amount
High-Pass Attenuate band below cutoff frequency	f_c - slope
Low-Pass Attenuate band above cutoff frequency	f_c - slope

band reject filter



lowpass filter



Technical focus

There is always a smooth transition between passband and stopband. It is often defined as the frequency at which the power transmitted by the filter drops to one half (in dB this is approximately –3 dB) of the maximum power transmitted in the passband.

Under special conditions a BPF may also be used as an LPF or as an HPF. An LPF can be simulated by setting the BPF's centre frequency to zero. The resulting cut-off frequency would in principle be one-half of the bw. As a consequence of this simulation, however, the cut-off frequency of the resulting low-pass is 70.7 per cent of the specified bandwidth, and not 50 per cent. For example, if the desired LPF cut-off frequency is to be 500 Hz, the bandwidth value for the BPF must be 1 kHz multiplied by 1.414 (that is, 1000 × 1.414). This is because the BPF is a two-pole filter; at its cut-off frequency (when output is 50 per cent) the output power of a true LPF of the same cut-off would be in fact 70.7 per cent.

An HPF can be made from a BPF by setting its centre frequency to be equal to the Nyquist frequency; i.e. the maximum frequency value produced by the system. HPFs made from BPFs suffer from the same approximation problems that affect LPFs.

The rate at which the attenuation increases is known as the slope of the filter – or rolloff. The rolloff is usually expressed as attenuation per unit interval, such as 6 dB per octave. In the stopband of an LPF with a 6 dB/octave slope, for example, every time the frequency doubles, the amount of attenuation increases by 6 dB. The slope of attenuation is determined by the order of the filter. Order is a mathematical measure of the complexity of a filter; in a digital filter, it is proportional to the number of calculations performed on each sample.