

ALIASING CANCELLATION IN AUDIO EFFECTS ALGORITHMS

Field of the Invention

This invention as it is defined in the appended claims is applicable in the field of Digital Audio Processing and, in particular, to aliasing cancellation in audio effects algorithm.

Background of the Invention

Audio effects such as delay, echo, reverberation, flanging and chorus are indispensable in music production and performance. Most of these effects today are implemented using digital signal processors. The modulating delay line forms the basis of many of these standard audio effects. The "dry" input is mixed with the effect signal, which is usually a linear function of the modulating delay.

Delay modulators introduce interpolation and aliasing artifacts. Previously, DSP processors were limited in their processing power and so it was understandable that algorithms made no extra effort to correct such errors. However, today, processing power is becoming lesser and lesser the limiting factor. In such a setting it is important that high quality be achieved by removing all artifacts and distortion.

The present invention seeks to cancel aliasing in audio effects algorithms.

Summary of the Invention

In accordance with the present invention, there is provided a method for effecting aliasing cancellation in an audio effects algorithm using a delay modulated signal, derived from interpolation of a delay modulator at an instantaneous sampling frequency, including:

determining the instantaneous sampling frequency $1/T_{\text{eff}}$; and

band limiting an input signal, to which the audio effects algorithm is to be applied,

- 2 -

to $\frac{1}{2} T_{lf}$ prior to interpolation.

Preferably, the delay modulated signal of a sampled version $x[n]$ of the band limited analogue signal $x(x)$ is expressed as $y[n] = x(\{n + D/2\sin(\omega_0 n)\}T_s)$

where $n = \text{integer}$

$D = \text{maximum delay}$

$\omega_0 = \text{oscillating frequency of the delay modulator; and}$

$T_s = 1/\text{sampling frequency.}$

Preferably, interpolation of the delay modulated signal $y[n]$ is computed using

$$y[n] = \sum_{k=-N}^N x[n' - k] \left[\frac{\sin[(\pi\Delta t - kT_s)T_{lf}]}{\pi(\Delta t - kT_s)T_s} \right]$$

$$n' = \text{INT}(n + D/2\sin(\omega_0 n))$$

$$\Delta t = \{n + D/2\sin(\omega_0 n)\}T_s - nT_s.$$

Preferably, the instantaneous sampling frequency, for a generalised modulating function $g(n)$ is derived from $T_{lf} = [1 + g(n) - g(n-1)]T_s$.

Detailed Description of a Preferred Embodiment

Consider a uniformly sampled version $x[n]$ of the bandlimited analog signal $x(t)$. A single reflection or echo of the signal can be implemented by the following filter, which adds to the input signal a delayed copy of itself

$$y[n] = ax[n] + (1 - a) * x[n + d(n)], \quad 1 > a > 0 \quad (1)$$

For an echo the delay $d(n) = D$, which essentially means that delay is constant. Note that although the equation (1) is *noncausal* the actual system can be made causal by adding an overall delay.

More interesting audio effects, such as flanging, chorus and reverberation can be created by allowing the delay $d(n)$ to vary in time e.g. $d(n) = D/2 \sin(\omega_0 n)$. Therefore, the delay varies oscillating between $-D/2$ and $D/2$, about the centre point. For the simple implementations of (1) the effect is like two singers - the lead singer sings normally while the other singer keeps racing up and down.

Consider the delay modulator (also known as vibrato) $y[n] = x[n + d(n)]$ in (1). It essentially attempts to perform non-uniform sampling of the signal $x(t)$ i.e.

$$y[n] = x(\{n + D/2 \sin(\omega_0 n)\}T_s)$$

Since the only information about $x(t)$ that is available to the system is that of values at discrete times nT_s , some form of interpolation is required to compute $y[n]$. The simplest approach is to

- 4 -

linear, cubic or lagrange interpolation $\sum_{k=-N}^N \left[x[n' - k] \left(\prod_{i=0, i \neq k}^N \frac{\Delta/T_s - i}{k - i} \right) \right]$,
 $n' = \text{INT}(n + D/2 \sin(w_o n))$ and $\Delta t = [n + D/2 \sin(w_o n)]T_s - nT_s$,

what these interpolation methods overlook is the aliasing resulting from sampling below the Nyquist limit.

Consider the instantaneous sampling frequency $(1/T_{\text{eff}})$ at time $nT_s + \Delta t$

$$\begin{aligned} T_{\text{eff}} &= \{n + D/2 \sin(w_o n)\}T_s - \{n - 1 + D/2 \sin(w_o(n - 1))\}T_s \\ &= \{1 + D/2 \sin w_o \cos(2n - 1)w_o\}T_s, \end{aligned}$$

To remove aliasing errors from the interpolated signal, the signal $x[n]$ should be first bandlimited to $1/2T_{\text{eff}}$. The interpolation and filtering however can be combined into one operation by using the analog reconstruction filter model. We now show how this is possible.

We know from sampling theorem that the signal $x(t)$, bandlimited to $F/2$, can be reconstructed from its samples $x(nT_s)$ as

$$x(t) = 1/F_s \int_{-F_s/2}^{F_s/2} \left[\sum_{n=-\infty}^{\infty} x[n] e^{-j2\pi n F_s t} \right] e^{j2\pi F_s t} dF$$

Since the new sampling frequency is $F_{\text{eff}} = 1/T_{\text{eff}}$ (and if $F_{\text{eff}} < F_s$) the limit in the above equation should be changed accordingly, thus leading to

$$x_c(t) = 1/F_s \int_{-F_{\text{eff}}/2}^{F_{\text{eff}}/2} \left[\sum_{n=-\infty}^{\infty} x[n] e^{-j2\pi n F_s t} \right] e^{j2\pi F_s t} dF$$

Upon simplification the above Eq leads to

$$x_c(t) = \sum_{n=-\infty}^{\infty} x[n] \left[\frac{\sin[(nT_s - t)/T_{\text{eff}}]}{\pi(nT_s - t)/T_s} \right]$$

Usually, the summation must be limited to a finite range, for practical implementation. Therefore

$y[n] = x_c(n'T_s + \Delta t)$, evaluated by considering $2N+1$ sample points $(x[n])$ about nT_s , is equivalent to

$$y[n] = \sum_{k=-N}^N x[n' - k] \left[\frac{\sin[(\pi \Delta t - kT_s)/T_{\text{eff}}]}{\pi(\Delta t - kT_s)/T_s} \right]$$

- 5 -

the above example the modulating function was the sinusoidal $\sin \omega_0 n$. If a general modulating function $g(n)$ is used the above formulation still works with the definition of the instantaneous sampling frequency as $(1/T_{\text{inf}})$ as $T_{\text{inf}} = (1 + g(n) - g(n-1))T_s$.

CLAIMS

1. A method for effecting aliasing cancellation in an audio effects algorithm using a delay modulated signal, derived from interpolation of a delay modulator at an instantaneous sampling frequency, including:

determining the instantaneous sampling frequency $1/T_{\text{eff}}$; and

band limiting an input signal, to which the audio effects algorithm is to be applied, to $\frac{1}{2} T_{\text{eff}}$ prior to interpolation.

2. A method as claimed in claim 1, wherein the delay modulated signal of a sampled version $x[n]$ of the band limited analogue signal $x(t)$ is expressed as

$$y[n] = x(\{n + D/2\sin(\omega_0 n)\}T_s)$$

where $n = \text{integer}$

$D = \text{maximum delay}$

$\omega_0 = \text{oscillating frequency of the delay modulator; and}$

$T_s = 1/\text{sampling frequency.}$

3. A method as claimed in claim 2, wherein interpolation of the delay modulated signal $y[n]$ is computed using

$$y[n] = \sum_{k=-N}^N x[n' - k] \left[\frac{\sin[(\pi \Delta t - k T_s)/T_{\text{eff}}]}{\pi(\Delta t - k T_s)/T_{\text{eff}}} \right]$$

$$n' = \text{INT}(n + D/2\sin(\omega_0 n))$$

$$\Delta t = [n + D/2\sin(\omega_0 n)]T_s - n'T_s$$

4. A method as claimed in any one of claims 1 to 3, wherein the instantaneous sampling frequency, for a generalised modulating function $g(n)$ is derived from

$$T_{\text{eff}} = [1 + g(n) - g(n-1)]T_s$$