

## Description

This invention relates to a tone signal processing device suitable for use in resampling at a lower rate a tone signal which has been sampled once with a sampling frequency of a relatively high rate.

An electronic musical instrument which has overcome the problem of an aliasing noise produced in the sampling process by harmonising the sampling frequency of a tone signal to be generated with the pitch of the tone signal is well known as a pitch synchronous type electronic musical instrument. An example of such a pitch synchronous type electronic musical instrument is disclosed in Japanese Patent Publication No. 171395/1982 (particularly Fig. 5).

Since, in this type of electronic musical instrument, the sampling frequency is different for each note, the frequency of a basic sampling clock used commonly for establishing the respective sampling frequencies must be the least common multiple of these sampling frequencies, which naturally is a fairly high frequency (e.g., 800 kHz).

The fact that the sampling frequency of a tone signal generated by a pitch synchronous type electronic musical instrument is of a high rate gives rise to the problem that the sampling frequency is too high when this device is applied to a device such as a digital effect device which operates with a clock frequency of a lower rate. Hence an arrangement is adopted so that a digital tone signal supplied at a high frequency is resampled at a lower frequency and the digital tone signal thus converted to a digital signal of lower rate frequency is applied to a digital effect circuit. With such an arrangement, however, there arises the problem that high frequency components contained in the original tone signal of the high-rate sampling frequency appear as an aliasing noise with respect to the low-rate sampling frequency.

DE-A-3226600 discloses, in Figure 24, a digital filter which allegedly serves to eliminate aliasing noise. Whilst this reference fails to disclose the detailed construction of the digital filter, it is believed that this will have the disadvantage that the tone signal outputted by the filter will be at a considerably lower sampling frequency, e.g. 12.5 kHz, than the sampling frequency of 50 kHz of the succeeding effect imparting device. Although it may be possible to match the two frequencies by resampling of the tone signal sample data at a clock frequency of 50 kHz, the precision of the output tone signal will be maintained at 12.5 kHz since the resampling will simply result in repetition of the same sample data. Accordingly the tone signal will be greatly degraded in quality by undue reduction of the sampling rate.

It is an object of the invention to provide an improved tone signal processing device for an electronic musical instrument.

According to the present invention, there is provided an electronic musical instrument comprising:

clock means for providing a relatively high frequency clock signal having an associated clock period;

tone signal generation means for providing a digital tone signal having a sampling period which corresponds to a sampling frequency which is  $1/N$  of the clock frequency, each cycle of said digital tone signal being formed by sampling at  $M$  sample points based on said sampling period; and

a digital filter for receiving said digital tone signal and filtering out substantially all frequencies above a predetermined cut-off frequency,

characterised in that said digital filter, comprises:

first, second and third delay circuits, coupled in series and providing first, second and third delayed digital tone signals, respectively, said first delay circuit receiving said digital tone signal, and each delay circuit having  $K$  stages, wherein  $K$  equals  $M/N$ , each of which delays said digital tone signal by one sampling period so that each said delay circuit delays said digital tone signal by  $K$  sampling periods;

a selector circuit coupled to said tone signal generation means, said first, second and third delay circuits and said clock means for selecting at each clock pulse one of said digital tone signal and said first, second and third delayed digital tone signals and providing such selected signal as a selector output signal;

filter coefficient storage and readout means for providing a predetermined filter coefficient signal at each clock period;

a multiplier circuit for receiving said selector output signal and said filter coefficient signal and multiplying said two signals at each clock period and providing the multiplied signal as a multiplier output signal;

an accumulator circuit for receiving said multiplier output signal and accumulating said multiplier output signal over  $M$  clock periods and providing an accumulated multiplier output signal every  $M$  periods; and in that said electronic musical instrument further comprises:

sampling means for receiving said accumulated multiplier output signal and resampling said signals every  $K$  sampling periods.

Thus the high frequency components which are likely to produce an aliasing noise with respect to the low-rate sampling frequency are removed from a digital tone signal by the digital filter. Accordingly the problem of the aliasing noise is eliminated in the case where a device using a relatively low clock rate such as a digital effect device is connected to a stage after the sampling means for imparting various tone effects.

The invention also provides a tone signal processing device comprising at least clock means for providing a clock signal having a relatively high frequency ( $\phi_0$ ) and having an associated clock period, tone signal generation means for providing a digital tone signal having a plurality of channels and having a first sampling period which corresponds to a first sampling frequency ( $\phi_1$ ) and based on which each cycle of said digital tone signal is formed by sampling at a specific number of sample points, and sampling means for resampling said digital tone signal at a second sampling frequency which is lower than said first sampling frequency ( $\phi_1$ ), characterised in that the device further comprises:

at least one delay means having a plurality of stages for receiving said digital tone signal and delaying said signal by a fixed multiple of said clock period at each stage and the or each delay means providing a delayed digital tone signal;

selector means for receiving said digital tone signal and said at least one delayed digital tone signal and alternately selecting at each clock period one of said digital tone signal and said at least one delayed digital tone signal and providing said selected digital signal as a selector output signal, wherein each of said digital tone signal and said at least one delayed digital tone signal are selected once during said first sampling period;

digital filter coefficient storage and readout means for providing a digital filter coefficient signal at each clock period;

multiplier means for receiving said selector output signal and said digital filter coefficient signal and providing a multiplied digital signal at each clock period; and

accumulator means for receiving the multiplied digital signal at each clock period and accumulating the received multiplied digital signals during a predetermined number of clock periods and providing the accumulated signal to said sampling means;

wherein said specific number of sample points per cycle corresponds to the ratio of the clock frequency to the second sampling frequency.

The invention further provides a tone signal processing method comprising providing a clock signal having a relatively high frequency ( $\phi_0$ ) and a clock period associated therewith, processing a digital tone signal having a first sampling period which corresponds to a first sampling frequency ( $\phi_1$ ), and resampling said digital tone signal at a second sampling frequency which is lower than said first sampling frequency, characterised in that said processing comprises:

delaying said digital tone signal in a plurality of consecutive stages by a fixed multiple of said first sampling period at each stage and providing a delayed digital tone signal every predetermined number of stages;

alternately selecting one of said digital tone signal and said delayed digital tone signal and providing said selected digital signal as a selector output signal, said selecting operation occurring at each clock period, and each of said digital tone signal and delayed digital tone signal being selected once during said first sampling period;

providing a digital filter coefficient signal at each clock period;

multiplying said selector output signal and said digital filter coefficient signal and providing a multiplied digital output signal at each clock period; and

accumulating said multiplied digital output signals during a predetermined number of clock periods and providing an accumulated digital signal for resampling at said second sampling frequency;

wherein said predetermined number of clock periods corresponds to the ratio of the clock frequency ( $\phi_0$ ) to said second sampling frequency.

In order that the invention may be more fully understood, a preferred embodiment of the invention will now be described, by way of example, with reference to the accompanying drawings, in which:

Fig. 1 is a block diagram showing an embodiment of the tone signal processing device according to the invention;

Fig. 2 is a block diagram showing the digital filter and the resampling device in Fig. 1 in detail;

Fig. 3 is a time chart showing an example of signals appearing in some parts of the circuit of Fig. 1;

Fig. 4 is a waveshape diagram showing an example of a digital tone signal supplied at a high-rate sampling frequency;

Fig. 5 is a diagram showing frequency component characteristics of the waveshape shown in Fig. 4;

Fig. 6 is a diagram showing low-pass filter characteristics realized by the digital filter of Fig. 2;

Fig. 7 is a diagram showing a waveshape obtained by filter controlling the waveshape of Fig. 4 with the low-pass filter characteristics of Fig. 6;

Fig. 8 is a diagram showing frequency component characteristics of the waveshape of Fig. 7;

Fig. 9 is a diagram showing frequency component characteristics of a waveshape obtained by resampling the waveshape of Fig. 7 with a low-rate sampling frequency; and

Fig. 10 is a diagram showing frequency component characteristics of a waveshape obtained by resampling the waveshape of Fig. 4 with a low-rate sampling frequency without a filtering process.

Referring to Fig. 1, tone signal generation means 1 generates tone signals corresponding to respective tone pitches (i.e., note names) in digital according to the pitch synchronizing system. Tones to be generated are designated by a keyboard (not shown) or other suitable means. This tone signal generation means 1 generates one or more digital tone signals of different tone pitches (note names) in a mixed state and, accordingly, the sampling frequency corresponds to the least common multiple of sampling frequencies which are synchronized in pitch with the respective tone pitches when these digital tone signals are viewed as a whole, which sampling frequency therefore is of a fairly high rate (e.g., 800 kHz). For such tone signal generation means 1 of the pitch synchronizing type, the device shown in the above-mentioned Japanese Preliminary Patent Publication No. 171395/1982 or Japanese Patent Application No. 2667/1984 (corresponding to EP 85100233.7) can be utilized. A digital tone signal produced by the tone signal generation means 1, i.e. a high-rate sampled wave signal, is supplied to a sound system 3 via a digital-to-analog converter 2 and sounded therefrom.

The digital tone signal provided by the tone signal generation means 1 is supplied also to a system including a digital effect imparting device 4. The digital effect imparting device 4 is a digital circuit for selectively imparting the digital tone signal with effects such as vibrato, chorus, ensemble and reverberation effects. Digital tone signals which are subjects of this digital effect imparting device 4 are of a relatively low-rate sampling frequency (e.g., 50 kHz). For such digital effect imparting device 4, the device shown in Japanese Preliminary Patent Publication No. 50595/1983 (corresponding to US-P4,472,993) or other suitable device can be used. The digital tone signal provided by the digital effect imparting device 4 is supplied to a sound system 6 via a digital-to-analog converter 5.

A resampling device 7 is provided between the tone signal generation means 1 and the digital effect imparting device 4 for converting the sampling frequency of the digital tone signal provided by the tone signal generation means 1 from a high-rate one (e.g., 800 kHz) to a low-rate one (e.g., 50 kHz). The digital tone signal which has been resampled with the low-rate sampling frequency in this resampling device 7 thereafter is applied to the digital effect imparting device 4.

A digital filter 8 is provided between the tone signal generation means 1 and the resampling device 7. This digital filter 8 filters the digital tone signal which are subjected to a high-rate sampling frequency with such filter characteristics as to be able to substantially remove an aliasing noise with respect to the low-rate sampling frequency (e.g., 50 kHz). As will be apparent from the sampling theorem, the aliasing noise occurs in the frequency region over 1/2 of the sampling frequency so that the filter characteristics of the digital filter 8 should preferably be set to a low-pass filter with a cut-off frequency equivalent to one half the low-rate sampling frequency for removing the aliasing noise.

A specific example of the device will be described hereunder on the assumption that the high-rate sampling frequency of 800 kHz and the low-rate sampling frequency of 50 kHz are used.

A sample value of the digital tone signal of the high-rate sampling frequency provided by the tone signal generation means 1 is designated by  $x_n$ . The suffix  $n$  represents a sample point number in one cycle of the tone signal which, by way of example, is any one of 0 through 63. A sample value of the digital tone signal provided by the digital filter 8 is designated by  $y_n$ . By way of example, the digital filter 8 is composed of an FIR filter (finite impulse response filter) of 64 stages having the following transfer function:

$$y_n = \sum_{i=0}^{63} h_i x_{n-i} \quad \dots \quad (1)$$

A sample value of the digital tone signal provided by the resampling device 7 is designated by  $Z_m$ . Since resampling device 7 converts the high sampling rate of 800 kHz to the low sampling rate of 50 kHz, the filter output signal  $y_n$  corresponding to the digital tone signal  $x_n$  sampled at the high sampling rate is resampled every 16 sample points. Accordingly,  $Z_m = y_{16m}$ .

Fig. 2 shows a specific example of the digital filter 8 and the resampling device 7. The digital filter 8 employs a single multiplier 9 on a time shared basis for multiplying filter coefficient  $h_i$  of each stage ( $i = 0$  to 63). Delay circuits 10, 11 and 12 each having 16 stages are cascade-connected. The delaying operation of these delay circuits are controlled with a sampling clock pulse synchronized with the high-rate sampling frequency of 800 kHz. The digital tone signal  $x_n$ , supplied in 16-bit parallel is applied to the first stage of the first delay circuit 10 and sequentially delayed by the sampling clock pulse  $\phi_1$  in synchronism with the high-rate sampling period. The digital tone signal  $x_n$ , which has not been delayed is applied to a "3" input of a selector 13, the output of the delay circuit 10 which has been delayed by 16 sampling periods is applied to a "2" input thereof, the output of the delay circuit 11 which has been delayed by 32 sampling periods is applied to a "1" input thereof and the output of the delay circuit 12 which has been delayed by 48 sampling periods is applied to a "0" input thereof. To a select control input of the selector 13 is applied a selection signal SEL. As shown in Fig. 3, this selection signal SEL successively changes between four states of "0" to "3" during one high-rate sampling period thereby successively selecting sample values of the digital tone signal applied to the "0" - "3" inputs. The state of the selection signal SEL changes in accordance with a clock pulse  $\phi_0$  having a frequency of 3.2 MHz which is four times as high as the high-rate sampling frequency.

Thus, the sample value  $x_n$  is selected by the selector 13 in a skipping manner every 16 sample points in accordance with the period of the clock pulse  $\phi_0$  and applied to the multiplier 9. The multiplier 9 receives at other input thereof a filter coefficient  $h_i$  read out from a coefficient ROM 14. A coefficient readout circuit 15 operates in response to the clock pulse  $\phi_0$  thereby designating the order  $i$  of the coefficient  $h_i$  to be read out at each period. The coefficient ROM 14 provides a coefficient  $h_i$  of the order  $i$  which has been designated by the coefficient readout circuit 15.

In the foregoing manner, each term  $h_i x_{n-i}$  of the above formula (1) is sequentially calculated every period of the clock pulse  $\phi_0$  in the multiplier 9. An accumulator 16 accumulates values of the respective terms  $h_i x_{n-i}$  supplied from the multiplier 9 in accordance with the clock pulse  $\phi_0$  to obtain the sum  $y$  of the series of the formula (1). Since  $i = 0$  to 63, the sum  $y_n$  of the formula (1) can be obtained by continuing the accumulation during 64 periods of the clock pulse  $\phi_0$ . A clear signal  $\overline{ACCLR}$  for the accumulator 16 becomes "0" every 64 periods of the clock pulse  $\phi_0$  as shown in Fig. 3 and clears contents of the accumulator 16 when it rises. The output of the accumulator 16 is applied to a latch circuit 17 which constitutes the resampling device 7. A latch pulse  $\overline{LP}$  of the latch circuit 17 is generated at a timing similar to that of the clear signal  $\overline{ACCLR}$ , latching contents of the accumulator 16 at its rising. Adjustment of the latch timing with the clear timing is made by a known technique so that the accumulator 16 is cleared after the contents of the accumulator 16 have surely been latched by the latch circuit 17. The low-rate sampling frequency of 50 kHz is used as frequencies of the latch pulse  $\overline{LP}$  and the clear signal  $\overline{ACCLR}$ .

The latch circuit 17 has a function of resampling the output tone signal of the digital filter 8 in accordance with the low-rate sampling frequency of 50 kHz and also a function of latching an accumulated value (a filter output value of one sample point) of the accumulator 17. As will be apparent from the foregoing description, the digital filter 8 performs a filter operation for one sample point by spending 64 periods of the clock pulse  $\phi_0$ , i.e., 16 periods of the high-rate sampling, i.e., one period of the low-rate sampling. Accordingly, the filter output is obtained not at each sample point of the high-rate sampling but every 16 sample points thereof in a skipping manner. No inconvenience, however, is caused by this arrangement, for the resampling in the latch circuit 17 has only to be performed in a skipping manner every 16 sample points and the filter output has only to be obtained at a sample point required for effecting this resampling. It is of course possible to obtain a filter output at each sample point by modifying the device in such a manner that speed of the time division operation of the digital filter 8 is increased or, conversely, providing plural multipliers 9 in correspondence to the respective delay stages and resample this filter output in a skipping manner in accordance with the low-rate sampling frequency.

For better understanding of the invention, an example of signals appearing in some parts of the circuit shown in Fig. 2 are shown in Fig. 3. In the figure, A, B, C and D represent sample values  $x_n$  or  $x_{n-i}$  of the tone signal applied to the "3", "0", "1" and "2" inputs of the selector 13 and E represents the sample value provided by the selector 13. H represents the coefficient  $h_i$  read out from the coefficient ROM 14 in correspondence to this E. G represents the output of the latch circuit 17, i.e., the filter controlled digital tone signal Z which has been converted to the low-rate sampling frequency.

For further understanding of the invention, examples of the waveshape and frequency characteristics of the tone signal are illustrated.

Fig. 4 is a waveshape diagram showing an example of the digital tone signal supplied in accordance with the high-rate sampling frequency of 800 kHz. Fig. 5 is a diagram showing frequency component characteristics of the waveshape shown in Fig. 4. In Fig. 5, all frequency components are not shown due to

the limitation in illustration and it should be understood that components exist even in a high frequency region over 100 kHz. Fig. 6 shows low-pass filter characteristics realized by the digital filter 8 consisting of an FIR filter of 64 stages with its cut-off frequency being set at 25 kHz. A waveshape obtained by passing the waveshape of Fig. 4 through the digital filter 8 of the low-pass filter characteristics of Fig. 6 is shown in Fig. 7. Frequency component characteristics of the waveshape of Fig. 7 are shown in Fig. 8 from which it will be noted that components below 25 kHz have substantially been cut off. Frequency component characteristics of a waveshape obtained by resampling the waveshape of Fig. 7 with the low-rate sampling frequency of 50 kHz are shown in Fig. 9. It will be seen from Fig. 9 that the waveshape has no aliasing noise but consists only of harmonic components. For the sake of comparison, frequency component characteristics of a waveshape obtained by resampling the waveshape of Fig. 4 at the low-rate sampling frequency of 50 kHz without using the filter are shown in Fig. 10. The black beard-like portions are crowded frequency components caused by an aliasing noise.

The digital filter employed in the present invention is not limited to the above described FIR filter of 64 stages but any type of filter including an FIR filter of other number of stages or an IIR filter (infinite impulse response filter) may be used.

The above embodiment has been described with respect to the example in which the high-rate sampling frequency of 800 kHz is converted to the fixed low-rate sampling frequency of 50 kHz. Relationship between the high-rate frequency and the low-rate one is not limited to this but other ratio may be selected as desired. The low-rate sampling frequency for the resampling is not limited to a fixed one but may be one which varies with time for producing a modulation effect.

The tone signal generation means is not limited to a polyphonic type device but a monophonic type device may also be employed. The invention is applicable not only to a device including the pitch synchronous type tone signal generation means but to any device in which the high-rate sampling frequency is converted to a low-rate sampling frequency.

According to the invention, a digital tone signal of a high-rate sampling frequency is converted to one of a low-rate sampling frequency after passing the digital tone signal through the digital filter 8 and an aliasing noise thereby can be removed with respect to the low-rate sampling frequency. Accordingly, the invention is useful in a case where a modulation effect device in which an input tone signal is required to be of a relatively low-rate sampling frequency is added to the electronic musical instrument, for coupling of the devices is realized without causing the problem of aliasing noise.

## Claims

### 1. An electronic musical instrument comprising:

clock means for providing a relatively high frequency clock signal ( $\phi_0$ ) having an associated clock period;

tone signal generation means (1) for providing a digital tone signal having a sampling period which corresponds to a sampling frequency ( $\phi_1$ ) which is  $1/N$  of the clock frequency ( $\phi_0$ ), each cycle of said digital tone signal being formed by sampling at  $M$  sample points based on said sampling period; and

a digital filter (8) for receiving said digital tone signal and filtering out substantially all frequencies above a predetermined cut-off frequency,

characterised in that said digital filter (8) comprises:

first, second and third delay circuits (10, 11, 12), coupled in series and providing first, second and third delayed digital tone signals, respectively, said first delay circuit (10) receiving said digital tone signal, and each delay circuit having  $K$  stages, wherein  $K$  equals  $M/N$ , each of which stages delays said digital tone signal by one sampling period so that each said delay circuit delays said digital tone signal by  $K$  sampling periods;

a selector circuit (13) coupled to said tone signal generation means (1), said first, second and third delay circuits (10, 11, 12) and said clock means for selecting at each clock pulse one of said digital tone signal and said first, second and third delayed digital tone signals and providing such selected signal as a selector output signal;

filter coefficient storage and readout means (14, 15) for providing a predetermined filter coefficient signal at each clock period;

a multiplier circuit (9) for receiving said selector output signal and said filter coefficient signal and multiplying said two signals at each clock period and providing the multiplied signal as a multiplier output signal;

an accumulator circuit (16) for receiving said multiplier output signal and accumulating said multiplier output signal over  $M$  clock periods and providing an accumulated multiplier output signal



the limitation in illustration and it should be understood that components exist even in a high frequency region over 100 kHz. Fig. 6 shows low-pass filter characteristics realized by the digital filter 8 consisting of an FIR filter of 64 stages with its cut-off frequency being set at 25 kHz. A waveshape obtained by passing the waveshape of Fig. 4 through the digital filter 8 of the low-pass filter characteristics of Fig. 6 is shown in Fig. 7. Frequency component characteristics of the waveshape of Fig. 7 are shown in Fig. 8 from which it will be noted that components below 25 kHz have substantially been cut off. Frequency component characteristics of a waveshape obtained by resampling the waveshape of Fig. 7 with the low-rate sampling frequency of 50 kHz are shown in Fig. 9. It will be seen from Fig. 9 that the waveshape has no aliasing noise but consists only of harmonic components. For the sake of comparison, frequency component characteristics of a waveshape obtained by resampling the waveshape of Fig. 4 at the low-rate sampling frequency of 50 kHz without using the filter are shown in Fig. 10. The black beard-like portions are crowded frequency components caused by an aliasing noise.

The digital filter employed in the present invention is not limited to the above described FIR filter of 64 stages but any type of filter including an FIR filter of other number of stages or an IIR filter (infinite impulse response filter) may be used.

The above embodiment has been described with respect to the example in which the high-rate sampling frequency of 800 kHz is converted to the fixed low-rate sampling frequency of 50 kHz. Relationship between the high-rate frequency and the low-rate one is not limited to this but other ratio may be selected as desired. The low-rate sampling frequency for the resampling is not limited to a fixed one but may be one which varies with time for producing a modulation effect.

The tone signal generation means is not limited to a polyphonic type device but a monophonic type device may also be employed. The invention is applicable not only to a device including the pitch synchronous type tone signal generation means but to any device in which the high-rate sampling frequency is converted to a low-rate sampling frequency.

According to the invention, a digital tone signal of a high-rate sampling frequency is converted to one of a low-rate sampling frequency after passing the digital tone signal through the digital filter 8 and an aliasing noise thereby can be removed with respect to the low-rate sampling frequency. Accordingly, the invention is useful in a case where a modulation effect device in which an input tone signal is required to be of a relatively low-rate sampling frequency is added to the electronic musical instrument, for coupling of the devices is realized without causing the problem of aliasing noise.

## Claims

### 1. An electronic musical instrument comprising:

clock means for providing a relatively high frequency clock signal ( $\phi_0$ ) having an associated clock period;

tone signal generation means (1) for providing a digital tone signal having a sampling period which corresponds to a sampling frequency ( $\phi_1$ ) which is  $1/N$  of the clock frequency ( $\phi_0$ ), each cycle of said digital tone signal being formed by sampling at M sample points based on said sampling period; and

a digital filter (8) for receiving said digital tone signal and filtering out substantially all frequencies above a predetermined cut-off frequency.

characterised in that said digital filter (8) comprises:

first, second and third delay circuits (10, 11, 12), coupled in series and providing first, second and third delayed digital tone signals, respectively, said first delay circuit (10) receiving said digital tone signal, and each delay circuit having K stages, wherein K equals  $M/N$ , each of which stages delays said digital tone signal by one sampling period so that each said delay circuit delays said digital tone signal by K sampling periods;

a selector circuit (13) coupled to said tone signal generation means (1), said first, second and third delay circuits (10, 11, 12) and said clock means for selecting at each clock pulse one of said digital tone signal and said first, second and third delayed digital tone signals and providing such selected signal as a selector output signal;

filter coefficient storage and readout means (14, 15) for providing a predetermined filter coefficient signal at each clock period;

a multiplier circuit (9) for receiving said selector output signal and said filter coefficient signal and multiplying said two signals at each clock period and providing the multiplied signal as a multiplier output signal;

an accumulator circuit (16) for receiving said multiplier output signal and accumulating said multiplier output signal over M clock periods and providing an accumulated multiplier output signal



every M periods; and in that said electronic musical instrument further comprises:

sampling means (17) for receiving said accumulated multiplier output signal and resampling said signals every K sampling periods.

- 5 2. A electronic musical instrument as claimed in claim 1, characterised in that said digital filter (8) is a finite impulse response filter.
3. A electronic musical instrument as claimed in claim 1 or 2, characterised in that it further comprises first and second sound systems (3, 6), one of the sound systems being coupled to said digital filter (8) and said sampling means (17) and the other sound system being coupled to said tone signal generation means (1), and digital effect imparting means (4) for digitally imparting a predetermined tone effect to the digital tone signal which has been resampled by said sampling means.
- 10 4. A tone signal processing device comprising at least clock means for providing a clock signal having a relatively high frequency ( $\phi_0$ ) and having an associated clock period, tone signal generation means (1) for providing a digital tone signal having a plurality of channels and having a first sampling period which corresponds to a first sampling frequency ( $\phi_1$ ) and based on which each cycle of said digital tone signal is formed by sampling at a specific number of sample points, and sampling means (17) for resampling said digital tone signal at a second sampling frequency which is lower than said first sampling frequency ( $\phi_1$ ), characterised in that the device further comprises:
  - 20 at least one delay means (10, 11, 12) having a plurality of stages for receiving said digital tone signal and delaying said signal by a fixed multiple of said clock period at each stage and the or each delay means providing a delayed digital tone signal;
  - 25 selector means (13) for receiving said digital tone signal and said at least one delayed digital tone signal and alternately selecting at each clock period one of said digital tone signal and said at least one delayed digital tone signal and providing said selected digital signal as a selector output signal, wherein each of said digital tone signal and said at least one delayed digital tone signal are selected once during said first sampling period;
  - 30 digital filter coefficient storage and readout means (14, 15) for providing a digital filter coefficient signal at each clock period;
  - multiplier means (9) for receiving said selector output signal and said digital filter coefficient signal and providing a multiplied digital signal at each clock period; and
  - 35 accumulator means (16) for receiving the multiplied digital signal at each clock period and accumulating the received multiplied digital signals during a predetermined number of clock periods and providing the accumulated signal to said sampling means (17);
  - wherein said specific number of sample points per cycle corresponds to the ratio of the clock frequency ( $\phi_0$ ) to the second sampling frequency.
- 40 5. A tone signal processing device as claimed in claim 4, characterised in that said sampling means (17) comprises a latch circuit.
6. A tone signal processing device as claimed in claim 4 or 5, characterised in that it comprises four delay means, wherein each said delay means includes 16 stages and each of said stages delays each digital tone signal by one clock period, and that said predetermined number of clock periods is 64.
- 45 7. A tone signal processing device as claimed in claim 6, characterised in that said first sampling period corresponds to 4 clock periods and wherein said fixed multiple of said clock period is 16.
8. A tone signal processing device as claimed in claim 4, 5, 6 or 7, characterised in that said clock frequency is 3.2 MHz, said first sampling frequency is 800 KHz, said second sampling frequency is 50 KHz, said specified number of sample points per cycle is 64 and said accumulated signal is provided at a frequency of 50 KHz.
- 50 9. A tone signal processing method comprising providing a clock signal having a relatively high frequency ( $\phi_0$ ) and a clock period associated therewith, processing a digital tone signal having a first sampling period which corresponds to a first sampling frequency ( $\phi_1$ ), and resampling said digital tone signal at a second sampling frequency which is lower than said first sampling frequency, characterised in that said processing comprises:
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delaying said digital tone signal in a plurality of consecutive stages by a fixed multiple of said first sampling period at each stage and providing a delayed digital tone signal every predetermined number of stages;

alternately selecting one of said digital tone signal and said delayed digital tone signal and providing said selected digital signal as a selector output signal, said selecting operation occurring at each clock period, and each of said digital tone signal and delayed digital tone signal being selected once during said first sampling period;

providing a digital filter coefficient signal at each clock period;

multiplying said selector output signal and said digital filter coefficient signal and providing a multiplied digital output signal at each clock period; and

accumulating said multiplied digital output signals during a predetermined number of clock periods and providing an accumulated digital signal for resampling at said second sampling frequency;

wherein said predetermined number of clock periods corresponds to the ratio of the clock frequency ( $\phi_0$ ) to said second sampling frequency.

## Revendications

1. Instrument de musique électronique comprenant une horloge pour fournir un signal de cadence ( $\phi_0$ ) d'une fréquence relativement élevée et ayant une période de cadence associée,

un moyen générateur de signaux sonores (1) pour fournir un signal sonore numérique ayant une période d'échantillonnage qui correspond à la fréquence d'échantillonnage ( $\phi_1$ ) qui est  $1/N$  de la fréquence de l'horloge ( $\phi_0$ ), chaque cycle dudit signal sonore numérique étant formé par l'échantillonnage en M points d'échantillonnage basé sur ladite période d'échantillonnage, et

un filtre numérique (8) pour recevoir ledit signal sonore numérique et en filtrer substantiellement toutes les fréquences au-delà d'une fréquence limite prédéterminée,

caractérisé en ce que ledit filtre numérique (8) comprend:

des premier, second et troisième circuits de retardement (10,11,12) couplés en série et fournissant des premier, second et troisième signaux sonores numériques retardés, respectivement, ledit premier circuit de retardement (10) recevant ledit signal sonore numérique et chaque circuit de retardement ayant K étages, où  $K = M/N$ , chacun desdits étages retardant le signal sonore numérique d'une période d'échantillonnage de sorte que chacun des circuits de retardement retarde ledit signal sonore numérique de K périodes d'échantillonnage,

un circuit sélecteur (13) couplé audit moyen générateur de signaux sonores (1), lesdits premier, second et troisième circuits de retardement (10,11,12) et ladite horloge pour choisir pour chaque impulsion de cadence un signal numérique dudit signal sonore numérique et desdits premier, second et troisième signaux sonores numériques retardés et fournissant ainsi un signal sélectionné comme signal de sortie du sélecteur,

un moyen de mémorisation et de lecture (14,15) à coefficient de filtre pour fournir un signal de coefficient de filtre prédéterminé à chaque période de cadence,

un circuit multiplicateur (9) pour recevoir ledit signal de sortie du sélecteur et ledit signal de coefficient de filtre et multiplier les deux signaux pour chaque période d'échantillonnage et fournir le signal multiplié comme un signal de sortie du multiplicateur,

un circuit accumulateur (16) pour recevoir ledit signal de sortie du multiplicateur et accumuler ledit signal de sortie du multiplicateur sur M périodes de cadence et fournir un signal de sortie multiplicateur accumulé toutes les M périodes, et

en ce que ledit instrument de musique électronique comprend en outre un moyen d'échantillonnage (17) pour recevoir ledit signal de sortie du multiplicateur accumulé et rééchantillonner lesdits signaux toutes les K périodes d'échantillonnage.

2. Instrument de musique électronique selon la revendication 1, caractérisé en ce que ledit filtre numérique (8) est un filtre à courbe de réponse aux impulsions limitée.

3. Instrument de musique électronique selon la revendication 1 ou 2, caractérisé en ce qu'il comprend en outre des premier et second signaux sonores (3,6), un des signaux sonores étant couplé audit filtre numérique (8) et audit moyen d'échantillonnage (17) et l'autre signal sonore étant couplé audit moyen générateur de signaux sonores (1), et un moyen donnant un effet numérique (4) pour donner numériquement un effet sonore prédéterminé au signal sonore numérique qui a été rééchantillonné par ledit moyen d'échantillonnage.



4. Dispositif de traitement de signal sonore, comprenant au moins une horloge pour fournir un signal de cadence ayant une fréquence ( $\varnothing_0$ ) relativement élevée et ayant une période de cadence y associée, un moyen générateur de signaux sonores (1) pour fournir un signal sonore numérique ayant une pluralité de canaux et ayant une première période d'échantillonnage qui correspond à la première fréquence d'échantillonnage ( $\varnothing_1$ ) et sur la base de laquelle chaque cycle dudit signal sonore numérique est formé en échantillonnant un nombre spécifique de points d'échantillonnage, et un moyen d'échantillonnage (17) pour rééchantillonner ledit signal sonore numérique à une seconde fréquence d'échantillonnage qui est inférieure à ladite première fréquence d'échantillonnage ( $\varnothing_1$ ), caractérisé en ce que ledit dispositif comprend en outre:
- 10 au moins un moyen de retardement (10,11,12) ayant une pluralité d'étages pour recevoir ledit signal sonore numérique et retarder ledit signal d'un multiple fixe de ladite période d'échantillonnage dans chaque étage et le ou chaque moyen de retardement fournissant un signal sonore numérique retardé,
- 15 un moyen sélecteur (13) pour recevoir ledit signal sonore et au moins un signal sonore numérique retardé et sélectionner alternativement à chaque période de cadence un signal du signal sonore numérique et du signal sonore numérique retardé et fournir ledit signal choisi comme signal de sortie du sélecteur, chacun du signal sonore numérique et du au moins un signal sonore numérique retardé sont choisis une fois pendant ladite première période d'échantillonnage,
- 20 un moyen de mémorisation et de lecture numérique à coefficient de filtre (14,15) pour fournir un signal numérique à coefficient de filtre pour chaque période de cadence,
- un moyen multiplicateur (9) pour recevoir ledit signal de sortie du sélecteur et ledit signal numérique à coefficient de filtre et fournir un signal numérique multiplié pour chaque période de cadence, et
- 25 un moyen accumulateur (16) pour recevoir le signal numérique multiplié à chaque période de cadence et accumuler les signaux numériques multipliés reçus pendant un nombre prédéterminé de périodes de cadence et fournir le signal accumulé audit moyen d'échantillonnage (17), ledit nombre spécifique de points d'échantillonnage par cycle correspondant au rapport de la fréquence de cadence ( $\varnothing_0$ ) à la seconde fréquence d'échantillonnage.
- 30 5. Dispositif de traitement de signal sonore selon la revendication 4, caractérisé en ce que ledit moyen d'échantillonnage (17) comprend un circuit de verrouillage.
- 35 6. Dispositif de traitement de signal sonore selon la revendication 4 ou 5, caractérisé en ce qu'il comprend en outre quatre moyens à retardement, chacun des moyens à retardement comprenant seize étages et chacun desdits étages retardant le signal sonore numérique d'une période de cadence et que ledit nombre prédéterminé de périodes de cadence est 64.
- 40 7. Dispositif de traitement de signal sonore selon la revendication 6, caractérisé en ce que ladite première période d'échantillonnage correspond à quatre périodes de cadence et que ledit multiple fixe de la période de cadence est 16.
- 45 8. Dispositif de traitement de signal sonore selon la revendication 4, 5, 6 ou 7, caractérisé en ce que ladite fréquence de cadence est 3,2 MHz, ladite première fréquence d'échantillonnage est 800 KHz, ladite seconde fréquence d'échantillonnage est 50 KHz, ledit nombre spécifique de points d'échantillonnage par cycle est 64 et que ledit signal accumulé est fourni à une fréquence de 50KHz.
- 50 9. Méthode de traitement de signal sonore comprenant les étapes de fournir un signal de cadence ayant une fréquence ( $\varnothing_0$ ) relativement élevée et une période de cadence y associée, de traiter un signal sonore numérique ayant une première période d'échantillonnage qui correspond à ladite première fréquence d'échantillonnage ( $\varnothing_1$ ), et rééchantillonner ledit signal sonore numérique à une seconde fréquence d'échantillonnage qui est inférieure à ladite première fréquence d'échantillonnage, caractérisée en ce que ladite étape de traiter comprend,
- 55 les étapes de retarder le signal sonore numérique dans une pluralité d'étages consécutives d'un multiple fixe de ladite première période d'échantillonnage dans chaque étage et fournir un signal sonore numérique retardé chaque nombre d'étages prédéterminé,
- de sélectionner alternativement un signal du signal sonore numérique et du signal sonore numérique retardé et fournir ledit signal numérique sélectionné comme signal de sortie sélecteur, ladite opération de sélection ayant lieu dans chaque période de cadence, et chacun du signal sonore



numérique et du signal sonore numérique retardé étant choisi une fois pendant ladite première période d'échantillonnage,

fournir un signal numérique à coefficient de filtre à chaque période de cadence,

multiplier ledit signal de sortie sélecteur et ledit signal numérique à coefficient de filtre et fournir un signal de sortie numérique multiplié à chaque période d'échantillonnage, et

accumuler lesdits signaux de sortie numériques multipliés pendant un nombre prédéterminé de périodes de cadence et

fournir un signal numérique accumulé pour rééchantillonner à ladite seconde fréquence d'échantillonnage, ledit nombre prédéterminé de périodes de cadence correspondant au rapport de la fréquence de cadence ( $\emptyset_0$ ) à ladite seconde fréquence d'échantillonnage.

### Patentansprüche

1. Elektronisches Musikinstrument mit einer Taktgebereinrichtung zum Bereitstellen eines Taktsignales ( $\emptyset_0$ ) relativ hoher Frequenz, das eine zugehörige Taktperiode hat,

eine Tonsignalgeneratoreinrichtung (1) zur Bereitstellung eines Digitaltonsignales, das eine Abtastperiode hat, welche der Abtastfrequenz ( $\emptyset_1$ ), entspricht, die dem  $1/N$  Teil der Taktfrequenz ( $\emptyset_0$ ) ist, wobei jeder Zyklus des Digitaltonsignales durch Abtasten an  $M$  Abtastpunkten auf Grund der Abtastperiode gebildet wird, und

ein Digitalfilter (8) zur Aufnahme des Digitaltonsignales und zum Herausfiltern von im wesentlichen allen Frequenzen oberhalb einer vorbestimmten Frequenz,

dadurch gekennzeichnet, dass der Digitalfilter (8) enthält:

erste, zweite und dritte in Reihe geschaltete Verzögerungskreise (10,11,12), welche jeweils erste, zweite und dritte verzögerte Digitaltonsignale bereitstellen, wobei der erste Verzögerungskreis (10) das Digitaltonsignal aufnimmt und jeder Verzögerungskreis  $K$  Stufen hat, wobei  $K = M/N$  ist, jede der Stufen das Digitaltonsignal um eine Abtastperiode verzögert, so dass jeder Verzögerungskreis das Tonsignal um  $K$  Abtastperioden verzögert.

ein Auswahl Schaltkreis (13), welcher an die Tonsignalgeneratoreinrichtung (1), den ersten, zweiten und dritten Verzögerungskreis (10,11,12) und die Taktgebereinrichtung angeschlossen ist, um bei jedem Taktimpuls eines der Digitaltonsignale und der ersten, zweiten und dritten verzögerten Digitaltonsignale auszuwählen und das so ausgewählte Signal als Auswahlgangssignal bereitzustellen,

eine Speicher- und Leseeinrichtung (14,15) mit Filterkoeffizient, um ein vorgegebenes Filterkoeffizientensignal bei jeder Taktperiode bereitzustellen,

ein Multiplikationskreis (9), welcher das Auswahlgangssignal und das Filterkoeffizientensignal empfängt und die beiden Signale bei jeder Abtastperiode multipliziert und das multiplizierte Signal als ein Multiplikationsgangssignal bereitzustellen,

ein Zwischenspeicherkreis (16) zum Empfangen des Multiplikationsgangssignals und zum Zwischenspeichern des Multiplikationsgangssignals über  $M$  Taktperioden und Bereitstellen eines zwischengespeicherten Multiplikationsgangssignals nach jeweils  $M$  Perioden, und

dadurch dass das elektronische Musikinstrument desweiteren eine Abtasteinrichtung (17) hat, um das zwischengespeicherte Multiplikationsgangssignal zu empfangen und diese Signale nach jeweils  $K$  Abtastperioden wieder abzutasten.

2. Elektronisches Musikinstrument nach Anspruch 1, dadurch gekennzeichnet, dass das Digitalfilter (8) ein Filter mit endlichem Impulsbereich ist.

3. Elektronisches Musikinstrument nach Anspruch 1 oder 2, dadurch gekennzeichnet, dass es desweiteren erste und zweite Tonsysteme (3,6) hat, wobei eines der Tonsysteme mit dem Digitalfilter (8) und der Abtasteinrichtung (17) gekoppelt ist und das andere Tonsystem mit der Tonsignalgeneratoreinrichtung (1) gekoppelt ist, so wie eine Digitaleffektzuteileinrichtung (4), um ein vorgegebenes Toneffekt dem Digitaltongenerator digital vorzugeben, welcher von der Abtasteinrichtung wieder abgetastet worden ist.

4. Tonsignalverarbeitungsvorrichtung mit wenigstens einem Taktgeber zum Bereitstellen eines Taktsignales, das eine relativ hohe Frequenz ( $\emptyset_0$ ) und eine zugeordnete Taktperiode hat, einer Tonsignalgeneratoreinrichtung (1) zum Bereitstellen eines Digitaltonsignales mit einer Vielzahl von Kanälen und einer ersten Abtastperiode, welche einer Abtastfrequenz ( $\emptyset_1$ ) entspricht und auf Grund deren jeder Zyklus des Digitaltonsignals durch Abtasten einer spezifischen Zahl von Abtastpunkten gebildet wird, und eine Abtasteinrichtung (17), um das Digitaltonsignal wieder bei einer zweiten Abtastfrequenz, welche kleiner



als die erste Abtastfrequenz ( $\emptyset_1$ ) ist, wieder abzutasten, dadurch gekennzeichnet, dass die Einrichtung desweiteren umfasst:

mindestens eine Verzögerungseinrichtung (10,11,12) mit einer Vielzahl von Stufen, um das Digitaltonsignal aufzunehmen und das Signal um ein festes vielfaches der Abtastperiode in jeder Stufe zu verzögern und die oder jede Verzögerungseinrichtung ein verzögertes Digitaltonsignal bereitstellt,

eine Auswahleinrichtung (13), um das Digitaltonsignal und das wenigstens eine verzögerte Digitaltonsignal zu empfangen und abwechselnd in jeder Abtastperiode vom dem Digitaltonsignal und dem wenigstens einen verzögerten Digitaltonsignal eines auszuwählen und das ausgewählte Digitaltonsignal als ein Auswahlausgangssignal bereitzustellen, wobei jedes von dem Digitaltonsignal und dem wenigstens einen verzögerten Digitaltonsignal einmal während der ersten Abtastperiode ausgewählt wird,

eine digitale Speicher- und Leseinrichtung (14,15) mit Filterkoeffizient, um ein Digitalfilterkoeffizientensignal während jeder Abtastperiode bereitzustellen,

eine Multiplikationseinrichtung (9) zur Aufnahme des Auswahlausgangssignales und des Digitalfilterkoeffizientensignals und Bereitstellen eines multiplizierten Digitalsignals bei jeder Abtastperiode, und

eine Zwischenspeichereinrichtung (16) zum Empfangen des multiplizierten Digitalsignals bei jeder Abtastperiode und Zwischenspeichern der empfangenen multiplizierten Digitalsignale während eine vorgegebenen Anzahl von Abtastperioden und Bereitstellen des zwischengespeicherten Signals an die Abtasteinrichtung (17),

wobei die spezifische Anzahl von Abtastpunkten pro Zyklus dem Verhältnis der Abtastfrequenz ( $\emptyset_0$ ) der zweiten Abtastfrequenz entspricht.

5. Tonsignalverarbeitungseinrichtung nach Anspruch 4, dadurch gekennzeichnet, dass die Abtasteinrichtung (17) einen Selbsthaltekreis enthält.

6. Tonsignalverarbeitungseinrichtung nach Anspruch 4 oder 5, dadurch gekennzeichnet, dass sie vier Verzögerungseinrichtungen enthält, wobei jede Verzögerungseinrichtung sechzehn Stufen hat, und jede der Stufen das Digitaltonsignal um eine Abtastperiode verzögert und, dass die vorgegebenen Anzahl von Abtastperioden 64 ist.

7. Tonsignalverarbeitungseinrichtung nach Anspruch 6, dadurch gekennzeichnet, dass die erste Abtastperiode vier Abtastperioden entspricht und, wobei das feste Vielfache der Abtastperiode 16 ist.

8. Tonsignalverarbeitungseinrichtung nach Anspruch 4, 5, 6 oder 7, dadurch gekennzeichnet, dass die Abtastfrequenz 3,2 MHz ist, die erste Abtastfrequenz 800 KHz ist, die zweite Abtastfrequenz 50 KHz ist, die spezifische Anzahl von Abtastpunkten pro Zyklus 64 ist und das zwischengespeicherte Signal bei einer Frequenz von 50 KHz bereitgestellt wird.

9. Tonsignalverarbeitungsverfahren, bei dem ein Abtastsignal mit einer relativ hohen Frequenz ( $\emptyset_0$ ) und einer dieser zugeordneten Abtastperiode, ein Digitaltonsignal, das eine erste Abtastperiode hat, die einer ersten Absatzfrequenz entspricht, verarbeitet wird und das Digitaltonsignal bei einer zweiten Abtastfrequenz, welche kleiner als die erste Abtastfrequenz ist, wieder abgetastet wird, dadurch gekennzeichnet, dass das Verarbeiten darin besteht,

das Digitaltonsignal in einer Vielzahl von aufeinanderfolgenden Stufen um ein festes Vielfaches der ersten Abtastperiode in jeder Stufe zu verzögern und ein verzögertes Digitaltonsignal nach jeder vorgegebenen Stufenzahl bereitzustellen,

abwechselnd ein Tonsignal aus dem Digitaltonsignal und dem verzögerten Digitaltonsignal auswählen und ein ausgewähltes digitale Signal als Auswahlausgangssignal bereitstellen, wobei der Auswahlbetrieb in jeder Abtastperiode erfolgt, und das Digitaltonsignal und das verzögerte Digitaltonsignal einmal während der ersten Abtastperiode ausgewählt wird,

ein Digitalfilterkoeffizientensignal während jeder Abtastperiode bereitstellen,

Multiplizieren des Auswahlausgangssignals und des Digitalfilterkoeffizientensignals und Bereitstellen eines multiplizierten Digitalausgangssignals während jeder Abtastperiode, und

Zwischenspeichern der multiplizierten Digitalausgangssignale, während einer vorgegebenen Anzahl von Abtastperioden und Bereitstellen eines zwischengespeicherten Digitalsignals zum Wiederabtasten bei einer zweiten Abtastfrequenz, wobei die vorgegebene Anzahl von Abtastperioden dem Verhältnis der Abtastfrequenz ( $\emptyset_0$ ) zur zweiten Abtastfrequenz entspricht.