

Description

This invention generally relates to a musical tone generating apparatus for use, for example, in an electronic musical instrument. The invention relates also to a musical tone data compression apparatus.

5 There have been proposed various kinds of signal processing methods in which an analog signal is first converted into sampled digital data to allow the analog signal to be digitally processed. Such methods are widely used particularly in tone generating apparatuses provided in, for example, voice synthesizers, electronic musical instruments and data communication systems.

Among such signal processing methods, a PCM (Pulse Code Modulation) method such as one
10 disclosed in U.S. Patent No. 4,383,462 is known. In this PCM method, an overall waveform of an analog signal such as a tone signal is sampled at a pretermined rate and the thus obtained series of sampled digital data are stored in a memory. And, the stored data are sequentially read from the memory and converted into analog signals to reproduce the analog signal. The PCM method is advantageous in that an exact reproduction of signal can be achieved and is therefore suitable for use in an electronic musical
15 instrument, but has such a deficiency that the amount of data to be stored in the memory is large.

To overcome the deficiency of the PCM method, various methods of reducing the amount of data, such as a DM (Delta Modulation) method, a DPCM (Differential Pulse Code Modulation) method and an ADPCM (Adaptive Differential Pulse Code Modulation) method, have been proposed. Even by these methods,
20 however, the amount of data can not sufficiently be reduced to such an extent that an electronic musical instrument can employ these methods to generate various musical tone signals. Also, in these methods, each data to be presently stored is formed based only on the precedingly obtained sample data and therefore includes unavoidable errors so that an exact reproduction of signal can not be achieved.

A musical tone generating apparatus in accordance with the prior art portion of claim 1 is known from US-A-4 133 241. This document discloses a musical instrument, having a memory in which weighting
25 coefficients (parameters) $a_1, a_2, \dots, a_K$ are stored. The waveshape amplitude at each sample point of the waveshape is computed on the basis of the weighting coefficients according to the formula

$$F(nT) = a_1 F((n-1)T) + a_2 F((n-2)T) + \dots$$

30 i.e. by effecting a linear operation on the sample points $F(n-1)$ previously calculated. This procedure serves to synthesize the musical wave shape in the musical instrument.

ICASSP 82, pages 614 to 617, discloses the derivation of an error signal between samples of original and synthetic speech for generating an excitation signal in an LPC system. This error signal is used in an "analysis-by-synthesis-procedure for determining the locations and the amplitudes of the (excitation)
35 pulses". The locations and amplitudes of the pulses are chosen to minimize the error.

It is an object of the invention to provide a tone generating apparatus in which the amount of data of a tone signal can be efficiently reduced and in which nevertheless an exact reproduction of a tone signal can be achieved.

It is further object of the invention to provide such a musical tone data compression apparatus in which
40 the amount of data of a tone signal such as a tone signal generated by a non-electronic musical instrument can be compressed and stored with a memory of a less capacity.

According to the present invention the musical tone generating apparatus is as claimed in claim 1.

The musical tone data compression apparatus according to the invention is mentioned in claim 13.

The memory means may alternatively store a series of second difference data each representing a
45 difference between a corresponding one of the series of difference data and that of the series of difference data produced a time interval corresponding to a period of the tone signal before the corresponding one of the series of difference data. In this case, the musical tone generating apparatus further comprises second data reproducing means which comprises first delay circuit means for delaying data inputted thereto by the time interval to output delayed data, and first adder means for adding the each second difference data
50 outputted from the memory means to the delayed data, each addition result being supplied to the first delay circuit means as the data and to the data reproducing means as the outputted difference data.

The memory means may alternatively store a series of third difference data each representing a difference between a corresponding one of the series of second difference data and that of the series of second difference data produced a second time interval corresponding to a predetermined number of
55 sampling intervals of the tone signal before the corresponding one of the series of second difference data. In this case, the musical tone generating apparatus further comprises third data reproducing means comprising second delay circuit means for delaying data inputted thereto by the second time interval to output delayed data, and second adder means for adding each of the third difference data outputted from

the memory means to the delayed data outputted from the second delay circuit means, each addition result being supplied to the second delay circuit means as the data and to the first adder means as the outputted second difference data.

Embodiments of the present invention will now be described with reference to the accompanying drawings, in which:-

- Fig. 1 is a block diagram of a data write section of a musical tone generating apparatus according to a first embodiment of the invention;
- Fig. 2 is a block diagram of a signal reproducing section of the first embodiment of the invention;
- Fig. 3 is a circuit diagram of the prediction data generating circuit 2 of the data write section of Fig. 1;
- Fig. 4 is an illustration showing the relation between the difference data $D(n)$ and $D(n+1)$ and the prediction data $FY(n)$ and $F(n+1)$ in the first embodiment of the invention;
- Fig. 5 is a block diagram of a data write section of a musical tone generating apparatus according to a second embodiment of the invention;
- Fig. 6 is a block diagram of a signal reproducing section of the second embodiment of the invention;
- Fig. 7 is an illustration showing the waveform of each of the input signal $F(t)$, the difference data $D(n)$ and the prediction data $E(n)$ in the second embodiment of the invention;
- Fig. 8 is an illustration showing the relation between the sampled data $F(n)$ and the prediction data $P(n)$ in the second embodiment of the invention;
- Fig. 9 is a circuit diagram of a modified form of the part of the data write section of Fig. 5;
- Fig. 10 is a time chart of the sampled data $F(n)$, the difference data $D(n)$ and the difference data $E(n)$ all obtained as the result of an experiment conducted on the circuit of Fig. 5;
- Fig. 11 is an illustration showing various values of the difference data $E(n)$ obtained as the result of an experiment conducted on the circuit of Fig. 5;
- Fig. 12 is a block diagram of an electronic keyboard musical instrument to which the first embodiment of the invention is applied;
- Fig. 13 is a block diagram of a data write section of a musical tone generating apparatus according to a third embodiment of the invention;
- Fig. 14 is a circuit diagram of the LPC 103 of the data write section of Fig. 13;
- Fig. 15 is a block diagram of a signal reproducing section of the third embodiment of the invention;
- Fig. 16 is a circuit diagram of the LPC demodulator 118 of the signal reproducing section of Fig. 15;
- Fig. 17 is a block diagram of a modified form of the data compression circuit 100c of the data write section of Fig. 13;
- Fig. 18 is a block diagram of a data write section of a musical tone generating apparatus according to a fourth embodiment of the invention;
- Fig. 19 is a block diagram of a signal reproducing section of the fourth embodiment of the invention;
- Fig. 20 is a block diagram of a data write section of a musical tone generating apparatus according to a fifth embodiment of the invention;
- Fig. 21 is a block diagram of a signal reproducing section of the fifth embodiment of the invention;
- Fig. 22 is a block diagram of the data compression circuit 110e of the data write section of Fig. 20;
- Figs. 23-(a) to 23-(c) are block diagrams of modified forms of the data compression circuit 100e of Fig. 22;
- Fig. 24 is a block diagram of a data write section of a musical tone generating apparatus according to a sixth embodiment of the invention;
- Fig. 25 is an illustration showing the variation of each of the tone signal $F1$ and the difference data $D1$ in the data write section of Fig. 24;
- Fig. 26 is a block diagram of a signal reproducing section of the sixth embodiment of the invention;
- Fig. 27 is a block diagram of a data write section of a musical tone generating apparatus according to a seventh embodiment of the invention;
- Fig. 28 is a block diagram of a signal reproducing section of the seventh embodiment of the invention;
- Fig. 29 is a circuit diagram of a further modified form of the PLPC 107; and
- Fig. 30 is a circuit diagram of a PLPC demodulator for the further modified PLPC of Fig. 29.

A musical tone generating apparatus provided in accordance with the present invention will now be described with reference to the accompanying drawings in which like reference characters denote corresponding parts in several views.

- Figs. 1 and 2 show a block diagram of a musical tone generating apparatus according to a first embodiment of the invention, wherein a data write section thereof for writing data representative of an analog signal into a memory M is shown in Fig. 1, and a signal reproducing section thereof for reproducing the analog signal from the data stored in the memory M is shown in Fig. 2.

In Fig. 1, shown at 1 is an analog-to-digital converter (hereinafter referred to as "ADC") which samples an amplitude of an analog input signal $F(t)$ such as a tone signal representative of a tone of an ordinary acoustic musical instrument at an interval defined by a clock signal ϕ and successively converts each of the sampled input signals $F(t)$ into a digital data or a digital sampled data $F(n)$. The sampled data $F(n)$ are successively outputted to a prediction data generating circuit 2 and a subtractor 3. The prediction data generating circuit 2 generates a prediction data $FY(n-1)$ based on a formula such as one given below and outputs it to the subtractor 3.

$$FY(n-1) = [F(n-2) + F(n)]/2 \quad (1)$$

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Fig. 3 shows a circuit diagram of the prediction data generating circuit 2 by way of example. In Fig. 3, shown at 4 is a register for storing the sampled data $F(n)$ in accordance with the clock signal ϕ . An output of the register 4 is therefore the sampled data $F(n-1)$ which is the sampled data generated a time period equal to one period of the clock signal ϕ (one clock time) before the presently generated sampled data or the current sampled data $F(n)$. The output of the register 4 is supplied to another register 5 of the same construction as the register 4. The sampled data $F(n)$ outputted from the ADC 1 is also supplied to a multiplier 6 which multiplies the sampled data $F(n)$ by a coefficient of "1/2" and supplies the result of this multiplication to an adder 7. An output of the register 5, i. e., the sampled data $F(n-2)$ is supplied to another multiplier 8 which multiplies the sampled data $F(n-2)$ by a coefficient of "1/2" and supplies the result of the multiplication to the adder 7. The adder 7 adds the outputs of the multipliers 6 and 8 together to obtain the prediction data $FY(n-1)$ as the result of the addition.

Assuming that the analog tone signal $F(t)$ has a waveform shown in Fig. 4, the ADC 1 outputs sampled data $F(n-1)$, $F(n)$, $F(n+1)$ and $F(n+2)$ at times $t(n-1)$, $t(n)$, $t(n+1)$ and $t(n+2)$, respectively. At the time $t(n+1)$, the prediction data generating circuit 2 adds the sampled data $F(n-1)$ to the sampled data $F(n+1)$, which is supplied thereto at this instant, and divide the result of this addition by "2" to produce the prediction data $FY(n)$ to be outputted. In a similar manner, at the time $t(n+2)$, the prediction data generating circuit 2 generates the prediction data $FY(n+1)$ based on the sampled data $F(n)$ and $F(n+2)$ and outputs the generated data. The subtractor 3 delays the sampled data $F(n-1)$ by a time period equal to one period of the clock pulse ϕ or one clock time and subtracts the prediction data $FY(n-1)$ from this delayed sampled data $F(n-1)$ to form a difference data $D(n-1)$. More specifically, the subtractor 3 calculates the difference data $D(n-1)$ based on the following formula (2) at the time $t(n)$ and outputs it.

$$D(n-1) = F(n-1) - FY(n-1) \quad (2)$$

As will be appreciated from Fig. 4, the difference data $D(n)$ is sufficiently smaller than the sampled data $F(n)$, so that the number of bit of the difference data $D(n)$ is also much less than that of the sampled data $F(n)$. And in this case, the prediction data generating circuit 2 and the subtractor 3 constitute a data compression circuit 100 of this system.

The difference data $D(n)$ thus obtained in the above-mentioned manner are successively stored into the memory M.

Referring now to Fig. 2, there is shown at 9 a sampled data reproduction circuit which reads the stored difference data $D(n-1)$ out of the memory M and reproduces the sampled data $F(n)$ in accordance with the read difference data $D(n-1)$. This reproduction circuit 9 will now be more fully described.

Formula (3) shown below is obtained by substituting the formula (1) for the formula (2).

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$$D(n-1) = F(n-1) - [F(n-2) + F(n)]/2$$

$$\text{or} \quad D(n) = F(n) - [F(n-1) + F(n+1)]/2 \quad \dots \quad (3)$$

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This formula (3) can be modified to obtain

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$$F(n+1) = 2[F(n) - D(n)] - F(n-1)$$

$$\text{or} \quad F(n) = 2[F(n-1) - D(n-1)] - F(n-2) \quad \dots \quad (4)$$

The sampled data reproduction circuit 9 reproduces the sampled data $F(n)$ based on the above formula (4). This reproduction operation of the circuit 9 will now be more specifically described on the assumption that first sampled data $F(0)$ and the second sampled data $F(1)$ of the sampled data $F(n)$ have previously been stored in the sampled data reproduction circuit 5. The reproduction circuit 9 first sequentially reads the sampled data $F(0)$ and $F(1)$ in accordance with the clock pulse ϕ , and then effects a calculation operation defined by the formula (4) on the read sampled data $F(0)$ and $F(1)$ and the difference data $D(1)$ read from the memory M. More specifically, the difference data $D(1)$ is subtracted from the data $F(1)$, and the result of this subtraction is multiplied by "2". Then, the data $F(0)$ is subtracted from the result of the multiplication to obtain the sampled data $F(2)$. After sequentially outputting the first and second sampled data $F(0)$ and $F(1)$, the reproduction circuit 9 outputs the thus reproduced sampled data $F(2)$ to a digital-to-analog converter (hereinafter referred to "DAC") 10. Then, after a lapse of one clock time of the clock pulses ϕ , the reproduction circuit 9 subjects the difference data $D(2)$ read from the memory M to the calculation operation defined by the formula (4) and outputs the sampled data $F(3)$ reproduced as the result of the calculation operation to the DAC 10. Thereafter, an operation similar to the above-described operation is repeated, and the sampled data $F(n)$ reproduced by the operation are successively supplied to the DAC 10. In this case, the sampled data reproduction circuit 9 constitutes a data expansion circuit 200 of this system. The DAC 10 converts each of the thus supplied sampled data $F(n)$ into an analog signal and outputs it. Thus, an analog signal identical with the inputted tone signal $F(t)$ is outputted from the DAC 10, although the outputted analog signal includes therein some quantization errors.

With this construction, the storage capacity of the memory M required to store the data representative of the tone signal $F(t)$ can be remarkably reduced in comparison with the conventional digital signal processing system of the type in which each sampled data is directly stored in a memory. Furthermore, the sampled data $F(n)$ can be reproduced by the sampled data reproducing circuit 9 in a simple manner.

A second embodiment of the invention will now be described.

Fig. 5 shows a block diagram of a data write section of the musical tone generating apparatus. This musical tone generating apparatus is so designed that it processes a periodic signal such as a tone signal representative of a tone of an acoustic musical instrument, and is improved so as to be superior to the system shown in Figs. 1 and 2 in the following two respects (1) and (2):

(1) It is well known that a tone signal representative of a tone of a musical instrument has a periodicity. The first improvement of the musical tone generating apparatus is made in view of the periodicity of the tone signal. With the musical tone generating apparatus shown in Figs. 1 and 2, the difference data $D(n)$ is stored in the memory M, whereas, with the second embodiment of this invention, a difference between the difference data $D(n)$ in the current period of the tone signal and the difference data $D(n)$ in the preceding period of the tone signal is stored in the memory. The difference between the above two difference data $D(n)$ is referred to as "difference data $E(n)$ ". It is now assumed that the tone signal $F(t)$ representative of a tone of a musical signal has a waveform shown in Fig. 7 and that a time $t(00)$ is the time when the tone begins to be generated. In this case, a difference between a difference data $D(K)_1$ obtained at a time $t(K)_1$ in the first period T_1 and a difference data $D(K)_2$ obtained at a time $t(K)_2$, which is identical in phase to the time $t(K)_1$, in the second period T_2 is extremely small. Also, a difference between the difference data $D(K)_2$ and a difference data $D(K)_3$ obtained at a time $t(K)_3$ is extremely small. In view of this fact, with the system shown in Figs. 5 and 6, the difference data $D(n)$ obtained by that sampling made one period of the tone signal $F(t)$ before the current sampling is subtracted from the difference data $D(n)$ obtained by the current sampling, and the result of this subtraction, that is, the difference data $E(n)$, is written into a memory M (see Fig. 7). In the first period T_1 of the tone signal $F(t)$, there is no preceding difference data $D(n)$, so that the difference data $D(n)_1$ is the difference data $E(n)_1$ to be written into the memory M.

Thus, the difference data $E(n)$ is much smaller than the difference data $D(n)$, so that the amount of data to be stored in the memory M in this embodiment becomes much smaller than that in the system shown in Figs. 1 and 2. Also, reproduction of the sampled data $F(n)$ from the difference data $E(n)$ can be established with a simple circuit as will be described later.

(2) With the apparatus shown in Figs. 1 and 2, the prediction data $FY(n)$ is calculated from the two sampled data $F(n-1)$ and $F(n+1)$ (refer to the formula (1)), whereas with the second embodiment of the invention a prediction data $FYa(n)$ is calculated from four sampled data $F(n-2)$, $F(n-1)$, $F(n+1)$ and $F(n+2)$. The calculation of the prediction data $FYa(n)$ will now be described with reference to Fig. 8.

First, three prediction data $P(n-1)$, $P(n)$ and $P(n+1)$ are calculated respectively from the sampled data $F(n-2)$ and $F(n-1)$, the sampled data $F(n-1)$ and $F(n+1)$, and the sampled data $F(n+1)$ and $F(n+2)$. These calculations are carried out based on the following three formulas:

$$P(n-1) = F(n-1) + [F(n-1) - F(n-2)] \quad (5)$$

$$P(n) = [F(n-1) + F(n+1)]/2 \quad (6)$$

$$P(n+1) = F(n+1) - [F(n+2) - F(n+1)] \quad (7)$$

The prediction data $P(n-1)$, $P(n)$ and $P(n+1)$ are multiplied respectively by weighting coefficients of "1", "2" and "1" and the results of these multiplications are summed. The sum is then divided by "4" to obtain the prediction data $FYa(n)$ as shown below.

$$FYa(n) = [P(n-1) + 2P(n) + P(n+1)]/4 \quad (8)$$

The foregoing is the basic concept of the calculation of the prediction data $FYa(n)$.

Next, by substituting the formulas (5) to (7) for the formula (8), the following formula is obtained:

$$FYa(n) = [-F(n-2) + 3F(n-1) + 3F(n+1) - F(n+2)]/4 \quad (9)$$

The prediction data $FYa(n)$ is calculated based on this formula (9).

Thus, according to this method of calculation, the prediction data $FYa(n)$ which is much closer in value to the actual sampled data $F(n)$ than the prediction data $FY(n)$ in the first embodiment can be obtained. And therefore, the number of bit or the value of the difference data $D(n)$ becomes much smaller. Incidentally, the aforesaid weighting coefficients can be determined by using, for example, the method of least squares so that the difference between the sampled data $F(n)$ and the prediction data $FYa(n)$ becomes minimum.

The construction of the musical tone generating apparatus according to the second embodiment of the invention will now be described with reference to Figs. 5 and 6. In Fig. 5, there is shown a data write section of the musical tone generating apparatus. The tone signal $F(t)$ representative of a tone of a musical instrument is converted into a digital sampled data $F(n)$ by a ADC 1 in accordance with a clock signal ϕ and supplied to a difference data generating circuit 11. The difference data generating circuit 11, which corresponds to the prediction data generating circuit 2 and subtractor 3 of the apparatus shown in Fig. 1, calculates the difference data $D(n)$ by subtracting the aforesaid prediction data $FYa(n)$ from the sampled data $F(n)$, as indicated below.

$$D(n) = F(n) - FYa(n) \quad (10)$$

By substituting the formula (9) for the above formula (10), the following formula is obtained:

$$D(n) = [F(n-2) - 3F(n-1) + 4F(n) - 3F(n+1) + F(n+2)]/4 \quad (11)$$

The difference data generating circuit 11 calculates the difference data $D(n)$ based on this formula (11) as described below and outputs it.

Shown at 12 to 16 are serially connected registers each composed of a predetermined number of D-type flip-flops (hereinafter referred to as "DFFs") which are triggered by the clock pulse ϕ . The sampled data $F(n)$ are sequentially inputted to the first register 12 and shifted from one register to another in accordance with the clock pulse ϕ . Outputs of the registers 12 to 16 are supplied to multipliers 17 to 21 to be multiplied by "1/4", "-3/4", "1", "-3/4" and "1/4", respectively. Outputs of the multipliers 17 to 21 are added together by an adder 22 and outputted therefrom as the difference data $D(n)$. Each of the multipliers 17 to 21 may be of a simple construction comprising a data shift circuit and adders. Thus, at the time $t(n+2)$, the sampled data $F(n+2)$, $F(n+1)$, $F(n)$, $F(n-1)$ and $F(n-2)$ are loaded respectively into the registers 12 to 16, so that the adder 22 outputs the difference data $D(n)$ obtained based on the formula (11). The construction of this difference data generating circuit 11 is identical to that of a linear-phase FIR filter.

The difference data $D(n)$ outputted from the adder 22 is supplied to a + input terminal of a subtractor 23. This subtractor 23 subtracts data at a - input terminal thereof from the data $D(n)$ supplied to the + input terminal thereof and outputs the result of this subtraction as a difference data $E(n)$. More specifically, this subtractor 23 is provided for subtracting that difference data $D(n)$ which was generated one period (one period of the tone signal $F(t)$) before, from the current difference data $D(n)$ supplied from the adder 22 and is fed at the - input terminal thereof with data outputted from a shift register 24 through an AND gate 25. The shift register 24 is provided for holding a series of difference data $D(n)$ for a time period equal to one

period of the tone signal $F(t)$ and comprises a plurality of stages equal in number to the samplings made during one period of the tone signal $F(t)$. The shift register 24 inputs the difference data $D(n)$ and shifts them from one stage to another in accordance with the clock pulse ϕ . The difference data $D(n)$ inputted to the first stage of the shift register 24 is therefore outputted from an output terminal Q of the last stage thereof when a time period equal to one period of the tone signal $F(t)$ has lapsed.

A signal IC shown in Fig. 5 is held in a "1" state during the first period T_1 of the tone signal $F(t)$, so that an inverter 27 outputs a "0" signal during the first period T_1 to cause the AND gate to output data of "0". As a result, the difference data $D(n)_1$ outputted from the adder 22 during the first period T_1 are outputted from the subtractor 23 as the difference data $E(n)_1$. These difference data $E(n)_1$ are sequentially written into the memory M and, at the same time, sequentially supplied to an adder 28. The adder 28 adds the difference data $E(n)_1$, i. e., the difference data $D(n)_1$, to the data "0" outputted from the AND gate 25 and sequentially supplies the results of the addition to the input terminal IN of the shift register 24, so that the difference data $D(n)_1$ are sequentially inputted to the shift register 24.

Then, the signal IC is rendered "0" at the beginning of the second period T_2 and held in the "0" state thereafter. When the signal IC is rendered "0", the AND gate 25 opens, so that the difference data $D(n)$ contained in the shift register 24 are supplied through the AND gate 25 to the - input terminal of the subtractor 23 as well as to the input terminal of the adder 28. As a result, during the second period T_2 , the subtractor 23 outputs difference data $E(n)_2$ defined by a formula given below.

$$E(n)_2 = D(n)_2 - D(n)_1 \quad (12)$$

The difference data $E(n)_2$ are sequentially written into the memory M. On the other hand, the adder 28 outputs the sum of the difference data $E(n)_2$ and the difference data $D(n)_1$ which are equal to the $D(n)_2$ as shown below.

$$E(n)_2 + D(n)_1 = D(n)_2 - D(n)_1 + D(n)_1 = D(n)_2 \quad (13)$$

And the difference data $D(n)_2$ are successively inputted to the shift register 24.

Thereafter, an operation similar to the above-described operation is repeated, and as a result the difference data $E(n)_1$, $E(n)_2$, ... are successively stored in the memory M.

The aforesaid difference data generating circuit 11, subtractor 23, shift register 24, AND gate 25, inverter 27 and adder 28 constitute a data compression circuit 100a of this embodiment. The circuit portion shown in the lower half of Fig. 5 may be modified as shown in Fig. 9. The circuit of Fig. 9 differs from that circuit portion shown in Fig. 5 in that the adder 28 is omitted.

Referring now to Fig. 6, there is shown a signal reproducing section of this musical tone generating apparatus. The difference data $E(n)$ read out of the memory M are supplied to one input terminal of an adder 31. The readout of the difference data $E(n)$ is carried out in accordance with the clock pulse ϕ . The adder 31 and a shift register 32 are provided for the purpose of reconstructing the difference data $D(n)$ from the read difference data $E(n)$. During the time when the difference data $E(n)_1$ corresponding to the first period T_1 , i. e., the difference data $D(n)_1$, are read from the memory M, the signal IC is in the state of "1", so that an inverter 33 outputs a "0" signal to cause an AND gate 34 to output data of "0". The data of "0" thus outputted from the AND gate 34 is supplied to the other input terminal of the adder 31. As a result, the difference data $D(n)_1$ are outputted from the adder 31 and supplied to a sampled data reproduction circuit 35 as well as to the shift register 32. This shift register 32 is identical in construction to the shift register 24 shown in Fig. 5 and sequentially stores therein the difference data $D(n)_1$ supplied from the adder 31. Each of the thus stored difference data $D(n)_1$ is shifted from one stage to another in accordance with the clock pulse ϕ and outputted from an output terminal Q of the last stage of the shift register 32 when a time period equal to one period of the tone signal $F(t)$ has lapsed. When the difference data $E(n)_2$ corresponding to the second period T_2 begin to be outputted from the memory M, the signal IC is rendered "0". This signal IC is kept in the "0" state thereafter, so that the AND gate 34 opens to sequentially supply the difference data $D(n)_1$ contained in the shift register 32 through the AND gate 34 to the other input terminal of the adder 31. As a result, the difference data $D(n)_2$ defined by a formula shown below are outputted from the adder 31.

$$E(n)_2 + D(n)_1 = D(n)_2 - D(n)_1 + D(n)_1 = D(n)_2 \quad (14)$$

The difference data $D(n)_2$ thus outputted from the adder 31 are supplied to the sampled data reproduction circuit 35 and also sequentially inputted to the shift register 32. And thereafter, an operation similar to the above operation is repeated, so that the difference data $D(n)_1$, $D(n)_2$, ... are sequentially supplied to the

sampled data reproduction circuit 35.

The sampled data reproduction circuit 35 then reproduces the sampled data $F(n)$ from the difference data $D(n)$. As mentioned before, the difference data $D(n)$ is formed based on the formula (11) which can be modified as follows:

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$$F(n+2) = -F(n-2) + 3F(n-1) - 4F(n) + 3F(n+1) + 4D(n) \quad (15)$$

The sampled data reproduction circuit 35 reproduces the sampled data $F(n)$ based on the above formula (15).

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The sampled data reproduction circuit 35 will now be more fully described. The sampled data reproduction circuit 35 comprises serially connected registers 36 to 39 each composed of a plurality of DFFs, multipliers 40 to 44 and an adder 45. It is assumed that the difference data $D(2)$ is now outputted from the adder 31. It is also assumed that the sampled data $F(0)$, $F(1)$, $F(2)$ and $F(3)$ are stored respectively in the registers 39, 38, 37 and 36 at this time, the sampled data $F(0)$ being the first one of the sampled data

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$F(n)$. In this case, the multipliers 44, 43, 42 and 41 multiply the sampled data $F(0)$, $F(1)$, $F(2)$ and $F(3)$ by coefficients of "-1", "3", "-4" and "3", respectively, and output the results of the respective multiplications to the adder 45. On the other hand, the multiplier 40 multiplies the difference data $D(2)$ by a coefficient of "4" and supplies the multiplication result to the adder 45. And therefore, when the difference data $D(n)$ is outputted from the adder 31, the adder 45 outputs the data shown below.

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$$-F(0) + 3F(1) - 4F(2) + 3F(3) + 4D(2) \quad (16)$$

This data is the sampled data $F(4)$, as will be appreciated from the formula (15). Thus, when the difference data $D(2)$ is outputted from the adder 31, the adder 45 outputs the sampled data $F(4)$ to a DAC 10 and the

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input terminal of the register 36. One clock time of the clock pulse ϕ later, the adder 31 outputs the difference data $D(3)$ whereupon the adder 45 outputs

$$-F(1) + 3F(2) - 4F(3) + 3F(4) + 4D(3) \quad (17)$$

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, i. e., the sampled data $F(5)$, since the sampled data $F(4)$ to $F(1)$ are stored respectively in the registers 36 to 39 at this time. In a similar manner, when the adder 31 sequentially outputs the difference data $D(4)$, $D(5)$, ..., the adder 45 sequentially outputs the sampled data $F(6)$, $F(7)$, ... to the DAC 10. Thus, an analog signal identical to the tone signal $F(t)$ of the tone of the musical instrument can be obtained at an output terminal of the DAC 10. The above-described sampled data reproduction circuit 35 is similar in construction

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to an IIR filter which is a kind of digital filter.

The adder 31, shift register 32, inverter 33, AND gate 34 and sampled data reproduction circuit 35 constitute a data expansion circuit 200a of this embodiment.

Although the prediction data $FYa(n)$ is obtained based on the precedingly sampled data $F(n-2)$ and $F(n-1)$ and the subsequently sampled data $F(n+1)$ and $F(n+2)$ in this embodiment, the prediction data $FYa(n)$ may be alternatively obtained based only on those sampled data which produced prior to the current sampled data.

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With the structure of the sampled data reproduction circuit 35 shown in Fig. 6, the first to fourth sampled data $F(0)$ to $F(3)$ are stored respectively in the registers 39 to 36 at the beginning of the reproduction of the tone signal $F(t)$. However, the registers 36 to 39 may alternatively be cleared at the

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beginning of the reproduction of the tone signal. In this case, although the tone signal with respect to the first several sampled data can not be reproduced accurately at the beginning of the reproduction operation, an accurate reproduction of the tone signal can be achieved with respect to the remaining sampled data.

With this embodiment, each difference data $E(n)$ is obtained as a difference between a difference data $D(n)$ in the current period of the tone signal and a corresponding difference data $D(n)$ in that period which is immediately before the current period. Alternatively, each difference data $D(n)$ may be obtained based on a difference between a difference data $D(n)$ in the current period and a corresponding difference data $D(n)$ in a period other than the period immediately before the current period. For example, each difference data $D(n)$ may be obtained based on a difference between a difference data $D(n)$ in the current period and a corresponding difference data $D(n)$ in that period which is two periods before the current period or half a

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period before the current period. In this case, the number of stage of each of the shift registers 24 and 32 must be equal to that of the sampling points which exist between the two difference data $D(n)$. Also, a memory such as a RAM may substitute for the shift registers 24 and 32 to delay the difference data $D(n)$ by the desired period of time.

It should also be noted that each of the first and second embodiments can be modified by adding a circuit to further reduce the amount of the data to be stored in the memory M. The circuit to be added may be a code converter which converts the difference data $D(n)$ or $E(n)$ into Shannon-Fano codes or Huffman codes to be stored into the memory M.

5 The Huffman code will now be briefly described.

In the first embodiment, values $(0, \pm 1, \pm 2, \dots)$ are generated as the difference data $D(n)$ at different probabilities. According to the Huffman encoding method, data having a higher probability of generation is assigned a code composed of fewer bits. The Shannon-Fano encoding method is similar in concept to the Huffman encoding method. Table 1 shows one example of the relationship between each value of the
10 difference data $D(n)$ and a corresponding Huffman code.

TABLE 1

Difference data $D(n)$	Probability of generation (%)	Huffman code HC	Number
0	50.3	1	1
-1	17.5	01	2
+1	10.6	001	3
others	9.3	0001	4+8
-2	4.5	00001	5
+2	3.5	000001	6
-3	2.2	0000001	7
+3	2.1	00000001	8

35 In Table 1, the measurement of the probability of generation is made with respect to a tone signal of a non-electronic musical instrument. The "others" in Table 1 designates any difference data $D(n)$ which is greater than +3 or less than -3, and the Huffman code corresponding to this difference data $D(n)$ is formed by adding a code of "0001" to the difference data $D(n)$ composed of, for example, eight bits.

40 In order to apply this Huffman encoding method to the first embodiment, an encoder H1 is interposed between the subtractor 3 and the memory M, as indicated by a broken line in Fig. 1, so that the difference data $D(n)$ are converted into the Huffman codes shown in Table 1 prior to being written into the memory M. Also, a decoder H2 is interposed between the memory M and the sampled data reproduction circuit 9, as indicated by a broken line in Fig. 2 to reproduce the difference data $D(n)$ from the Huffman codes read from
45 the memory M. The Huffman encoding method can also be applied to the second embodiment in a similar manner as shown in Figs. 5 and 6. The application of the Huffman encoding method reduces the amount of the data to be stored in the memory M by about three-eighths times.

The result of an experiment made on the circuit of Fig. 5 will now be described.

50 Figs. 10-(a) shows the variation of 12-bit sampled data $F(n)$ of a tone signal $F(t)$ representative of a trumpet tone (8' -G4; 392Hz) of a pipe organ, wherein the tone signal $F(t)$ is sampled at an interval of clock signal ϕ , i. e., at a sampling rate of 35 KHz. And, Figs. 10-(b) and 10-(c) respectively show difference data $D(n)$ and difference data $E(n)$ of the sampled data $F(n)$ shown in Fig. 10-(a). As will be appreciated from Figs. 10-(a) to 10-(c), the magnitude of the difference data $D(n)$ is much smaller than that of the sampled data $F(n)$, and the magnitude of the difference data $E(n)$ from the second period thereof is much smaller
55 than that of the difference data $D(n)$. Fig. 11 shows how many times each value of the difference data $E(n)$ is generated during the first 512 samples of the tone signal $F(t)$. For example, the difference data $E(n)$ of "0" is generated 33 times, the difference data $E(n)$ of "1" 40 times, the difference data $E(n)$ of "2" 34 times, the difference data $E(n)$ of "-1" 58 times, and the difference data $E(n)$ of "-2" 28 times. In Fig. 11, those of

the values of the difference data $E(n)$ which are greater than "+11" or less than "-11" are generated only during the first period of the tone signal $F(t)$, and those difference data $E(n)$ of such large values are hardly generated from the second period of the tone signal $F(t)$. The reason why the difference data $E(n)$ of such large values are generated during the first period of the tone signal $F(t)$ is that the difference data $D(n)$ are used as the difference data $E(n)$ during the first period of the tone signal $F(t)$.

In the aforesaid second embodiment, the difference data generating circuit 11 shown in Fig. 5 is so arranged that the registers 12 to 16 are initially cleared. And the reason why the trumpet tone of a pipe organ is used in the experiment is that the trumpet tone contains a lot of higher harmonic components and has a complicated waveform.

A musical tone generating apparatus to which the first embodiment of the present invention is applied will now be described with reference to Fig. 12.

As shown in Fig. 12, the musical tone generating apparatus comprises the memory M in which difference data $D(n)$ of an overall waveform of each of a plurality of tone signals $F(t)$ are stored. In this case, difference data $D(n)$ of a plurality of tone signals $F(t)$ of different tone pitches are stored per each of different tone colors such as a piano tone and a flute tone. When a desired one of the tone colors is selected at a tone-color selection section 50, the tone-color selection section 50 outputs data TC to the memory M to designate a memory area thereof where the difference data $D(n)$ of the tone signals $F(t)$ of the selected tone color are stored. When a key is depressed at a keyboard 51, a key-depression detection section 52 detects the depression of the key and outputs a key code KC representative of the depressed key together with a key-on signal KON. The key-on signal KON is kept in a "1" state during the depression of the key. An address generator 53 converts the key code KC fed from the key-depression detection section 52 into a corresponding address data AD1 and outputs it to the memory M. The address generator 53 also outputs to the memory M, from the leading edge of the key-on signal KON, address data ADD of "0" which is incremented by one at a predetermined time interval thereafter. When the address data AD1 is supplied to the memory M, a region in that memory area of the memory M designated by the data TC is selected, the region in the memory area of the memory M corresponding to the address data AD1. The thus selected region stores the difference data $D(n)$ of that of the tone signals $F(t)$ which is of the tone color designated by the tone-color selection section 50 and of the tone pitch designated by the key code KC. And, when the address data ADD is supplied to the memory M, the difference data $D(n)$ are sequentially read out of the addresses of the selected region from its relative address "0" or the first address thereof, and supplied to the sampled data reproduction circuit 9a. As a result, the sampled data $F(n)$ are sequentially outputted from the sampled data reproduction circuit 9a and converted into an analog tone signal $F(t)$ by the DAC 10. The analog tone signal $F(t)$ thus obtained is supplied to a sound system 54 which in turn amplifies the tone signal $F(t)$ to drive a loudspeaker (not shown) to thereby produce a musical tone.

When it is desired to apply a variation of pitch such as "vibrato" to the produced musical tone, the intervals of the output of the address data ADD may be changed. In this case, the sampled data reproduction circuit 9a carries out the processing of data in synchronism with the intervals of the output of the addressed data ADD. Although the memory M stores a plurality of groups of difference data $D(n)$ of different tone pitches per each of different tone colors in the aforesaid apparatus, the memory M may alternatively store only one group of difference data $D(n)$ per each of the different tone colors. Also, the memory M may alternatively store one group of difference data $D(n)$ per a predetermined number of different tone pitches with respect to each of the different tone colors. In this case, the address generator 53 generates the address data ADD at a time interval determined by the key code KC.

A third embodiment of the invention will now be described with reference to Figs. 13 to 16.

In Fig. 13, a periodic analog tone signal F1 is supplied through an input terminal T1 to the ADC 1 which converts the tone signal F1 into a digital form at a predetermined sampling rate to produce a sampled digital data F2. The sampled data F2 outputted from the ADC 1 is supplied to a subtractor 102 and a linear predictive coding circuit (hereinafter referred to as "LPC") 103. In this case, the number of samples made by ADC 1 during one period of the tone signal F1 is set to "m". The LPC 103 calculates a prediction data FY1 of the sampled data F2 using a well-known linear prediction method and comprises such a circuit as that shown in Fig. 14. In Fig. 14, blocks 110 represent delay circuits of the same construction each of which functions to delay input data thereof by a time period equal to one sampling time of the tone signal F1, as identified by "Z-1". Each delay circuit 110 comprises, for example, a register composed of a predetermined number of D-type flip-flops and an output thereof is supplied to a corresponding one of multipliers 111 which are equal in number to the delay circuits 110. The multipliers 111 multiply inputs thereof respectively by coefficients $a_1, a_2, \dots, a_p$, and outputs of these multipliers 111 are added together by an adder 113 to obtain the prediction data FY1. With this LPC 103, the prediction data FY1 is calculated based only on the precedingly obtained sampled data. However, the LPC 103 may alternatively calculate the prediction data

FY1 based on the precedingly obtained sampled data and the subsequently obtained sampled data. In this case, a delay circuit need be interposed between the ADC 1 and the subtractor 102, as indicated by a broken line in Fig. 13. The subtractor 102 subtracts the output of the LFC 103, i. e., the prediction data FY1, from the output F2 of the ADC 1 and outputs the result of this subtraction as a first difference data D1. As the difference data D1 represents the difference between the sampled data F2 and the prediction data FY1, the difference data D1 is far smaller than the sampled data F2. Thus, the number of bit of the difference data D1 can be reduced to achieve a compression of information. In Fig. 13, that portion of the circuit which comprises the aforesaid subtractor 102 and LFC 103 and is denoted by a reference numeral 104 is hereinafter referred to as "LPCC".

10 The first difference data D1 outputted from the subtractor 102 is supplied to a delay circuit 105 which delays the first difference data D1 by a time period equal to one period of the tone signal F1, i. e., a time period corresponding to m sampling times as identified by "Z^m". The delay circuit 105 may comprise a m-stage shift register. An output of this delay circuit 105 is supplied to a subtractor 106 which subtracts the output of the delay circuit 105 from the output D1 of the subtractor 102 and outputs the result of the subtraction as a second difference data D2. As described above, since the tone signal F1 is a periodic signal, the first difference data D1 also periodically varies in synchronism with the tone signal F1. And therefore, the difference between the presently produced first difference data D1 and the first difference data D1 produced m sampling times before, that is to say, the second difference data D2, is rendered much smaller than the first difference data D1. In other words, the number of bit of the second difference data D2 is smaller than that of the first difference data D1. In Fig. 13, that portion of the circuit which comprises the aforesaid delay circuit 105 and subtractor 106 and is denoted by a reference numeral 107 is hereinafter referred to as "PLPC (periodic linear predictive coding circuit)".

The second difference data D2 thus outputted from the subtractor 106 is supplied to a delay circuit 108 which delays the second difference data D2 by one sampling time and outputs the delayed data to a subtractor 109. The delay circuit 108 comprises, for example, a register composed of D-type flip-flops driven by the clock signal ϕ . The subtractor 109 subtracts the output of the delay circuit 108 from the second difference data D2 and outputs the result of the subtraction to an output terminal T2. In Fig. 13, that portion of the circuit which comprises the aforesaid delay circuit 108 and subtractor 109 and is denoted by a reference numeral 110 is a well-known modulation circuit based on the differential pulse code modulation method and hereinafter referred to as "DPCM". The aforesaid LPCC 104, PLPC 107 and DPCM 110 constitute a data compression circuit 100c of this embodiment.

With the above arrangement, a compression of data is effected at each of the LPCC 104, PLPC 107 and DPCM 110, so that the number of bit of the data obtained at the output terminal T2 is much smaller than that of the sampled data F2 at the output terminal of the ADC 1. Each of the circuit components of the circuit of Fig. 13 may alternatively be constructed by an analog circuit. In this case, the ADC 1 may be interposed between the DPCM 110 and the output terminal T2 so that an ADC of fewer output bits can be used. The data thus obtained at the output terminal T2 is stored into the memory M or transmitted to a remote terminal (not shown) through a transmission line.

Fig. 15 shows a circuit for reproducing the sampled data F2 from the output data of the circuit of Fig. 13. In Fig. 15, an input terminal T3 is supplied with a series of data which are identical in value and sequence to the data outputted from the output terminal T2 of the data write section of Fig. 13. The respective data of the series of data are sequentially supplied to the input terminal T3 at a time interval equal to that of the samplings made at the ADC 1. The data supplied to the input terminal T3 is fed to an adder 112 whose output is supplied to a delay circuit 113 of the same construction as the delay circuit 108 of Fig. 13. An output of this delay circuit 113 is supplied to the adder 112 and is added thereby to the data fed to the input terminal T3. The adder 112 and the delay circuit 113 constitute a DPCM demodulator 114 for producing a first reproduction data R1 which is identical to the aforesaid second difference data D2. The first reproduction data R1 is supplied to another adder 115 whose output is supplied to a delay circuit 116 of the same construction as the delay circuit 105 of Fig. 13. An output of the delay circuit 116 is supplied to the adder 115 and is added thereby to the first reproduction data R1. The aforesaid adder 115 and delay circuit 116 constitute a PLPC demodulator 117 for producing a second reproduction data R2 which is identical to the first difference data D1. The second reproduction data R2 is supplied to an LPC demodulator 118 which converts the second reproduction data R2 into a third reproduction data R3 which is identical to the sampled data F2. The LPC demodulator 118 has such a construction as that shown in Fig. 16. In Fig. 16, the second reproduction data R2 is supplied to an adder 119 whose output is fed to an input terminal of a circuit 103a of the same construction as that of the LFC 103 of Fig. 13. An output of this circuit 103a is supplied to the adder 119 to be added to the second reproduction data R2 to thereby produce the third reproduction data R3 or the reproduced sampled data F2. The sampled data F2 thus reproduced is fed

to the DAC 10 which in turn converts the sampled data F2 into an analog signal to obtain the reproduced tone signal F1. The aforesaid DPCM demodulator 114, PLPC demodulator 117 and LPC demodulator 118 constitute a data expander circuit 200c of this embodiment.

The data compression circuit 100c shown in Fig. 13 comprises only one DPCM 110, however the circuit 100c may be modified to comprise a plurality of DPCM 110 connected in series to achieve further compression of data as shown in Fig. 17. It is apparent that the circuit 200c of Fig. 15 must be modified to have serially connected DPCM demodulators 114 equal in number to the DPCM 110 in this case. Also, each of the delay circuits 105 and 116 may be modified such that data inputted thereto is delayed by 2m sampling times (a period of time equal to two periods of the tone signal $F(t)$), 3m sampling times (a period of time equal to three periods of the tone signal $F(t)$), or more. Furthermore, each of the delay circuits 108 and 113 may be modified such that data inputted thereto is delayed by two sampling times, three sampling times or more.

A fourth embodiment of the invention will now be described with reference to Figs. 18 and 19.

In Fig. 18, the ADC 1, PLPC 107 and LPCC 104 are serially connected in this order between the input terminal T1 and the output terminal T2. The sampled data F2 outputted from the ADC 1 is first converted into a first difference data D11 by the PLPC 107 and then further converted into a second difference data D12 by the LPCC 104. The thus obtained second difference data D12 is outputted from the output terminal T2 and is written into the memory M or transmitted to a remote terminal. With this arrangement, the sampled data F2 is compressed by the PLPC 107 prior to being supplied to the LPCC 104. And therefore, the number of bit of the data on which the LPCC 104 effects an operation becomes smaller, so that the circuit of the LPCC 104 can be simplified. A data compression circuit 100d of this embodiment is thus constituted by the PLPC 107 and the LPCC 104.

Fig. 19 shows a signal reproducing section for reproducing the tone signal F1 from the output data D12 of the data write section of Fig. 18. In Fig. 19, the LPC demodulator 118, the PLPC demodulator 117 and the DAC 10 are serially connected in this order between the input terminal T3 and the output terminal T4. The data D12 supplied to the input terminal T3 is first converted by the LPC demodulator 118 into a first reproduction data R11 which is identical to the data D11 shown in Fig. 18. The first reproduction data R11 is then converted by the PLPC demodulator 117 into a second reproduction data R12 which is identical to the sampled data F2 shown in Fig. 18. The thus obtained second reproduction data R12 is further converted into an analog form by the DAC 10 to obtain the reproduced tone signal F1 which is taken from the output terminal T4. The LPC demodulator 118 and the PLPC demodulator 117 constitute a data expander circuit 200d of this fourth embodiment.

A fifth embodiment of the invention will now be described with reference to Figs 20 and 21.

The tone signal F1 applied to the input terminal T1 is converted by the ADC 1 into the sampled data F2 which is then compressed by the LPCC 104 to obtain a first difference data D21. This first difference data D21 is further compressed by a PLPC 122 to obtain a second difference data D22 which is supplied to the output terminal T2. The PLPC 122 is similar to the PLPC 107 of Fig. 13 but differs therefrom in that a multiplier 123 is connected to the input terminal of the delay circuit 105 so that the first difference data D21 is multiplied by a coefficient A prior to being supplied to the delay circuit 105. The coefficient A has a value of near "1" and is set to such a value that allows an efficient compression of data to be effected at the PLPC 122. The reason why the first difference data D21 is multiplied by the coefficient A will now be described. It is assumed here that the first difference data D21 increases at a constant rate. In this case, if the first difference data D21 is multiplied by a coefficient corresponding to the rate of increase thereof prior to application to the delay circuit 105, the difference between the current difference data D21 and the output of the delay circuit 105 becomes smaller. As a result, the efficiency of compression of data is improved. Thus, the coefficient A may be determined in accordance with the rate of variation of the difference data D21, and it is preferable that the coefficient A be varied with the lapse of time. In Fig. 20, the multiplier 123 may alternatively be connected to the output terminal of the delay circuit 105. In this case, if the second difference data D22 obtained at the output terminal T2 is stored into a memory, the coefficient A need be stored into the memory simultaneously. A data compression circuit 100e of this fifth embodiment is constituted by the LPCC 104 and the PLPC 122.

Fig. 21 shows a signal reproducing section of this embodiment for reproducing the tone signal F1 from the data D22 compressed by the circuit shown in Fig. 20. In Fig. 21, the data supplied to the input terminal T3 is converted by a PLPC demodulator 125 into a first reproduction data R21 which is identical to the aforesaid first difference data D21. The PLPC demodulator 125 is similar to the PLPC 117 of Fig. 15 but differs therefrom in that a multiplier 126 is connected to the output terminal (or the input terminal) of the delay circuit 116 so that the output (or the input) of the delay circuit 116 is multiplied by the coefficient A prior to application to the adder 115. The first demodulation data R21 outputted from the PLPC demodulator

125 is then converted by the LPC demodulator 118 into a second reproduction data R22 which is identical to the sampled data F2. The thus obtained second reproduction data R22 is further converted by the DAC 10 into an analog form to obtain the reproduced tone signal F1 which is taken from the output terminal T4.

Modified forms of the above fifth embodiment will now be described.

- 5 Fig. 22 again shows that portion of the circuit of Fig. 20 disposed downstream of the ADC 1. In Fig. 22, block 130 represents the LPC 103 of Fig. 20 and block 131 represents the combination of the multiplier 123 and the delay circuit 105. $P_1(z)$ and $P_2(z)$ shown respectively in the blocks 130 and 131 are transfer functions thereof and can be expressed as follows:

$$10 \quad P_1(z) = \sum_{n=1}^k a_n z^{-n} \quad \dots \quad (18)$$

$$15 \quad P_2(z) = A Z^{-m} \quad (19)$$

An overall transfer function of the circuit shown in Fig. 22 is therefore expressed as:

$$20 \quad \begin{aligned} & [1 - P_1(z)][1 - P_2(z)] \\ & = 1 - P_1(z) - P_2(z) + P_1(z) P_2(z) \quad \dots \quad (20) \end{aligned}$$

$$= 1 - P_1(z) - P_2(z)[1 - P_1(z)] \quad \dots \quad (21)$$

$$25 \quad = 1 - [P_1(z)[1 - P_2(z)] + P_2(z) \quad \dots \quad (22)$$

Figs. 23-(a), 23-(b) and 23-(c) show circuits formed based respectively on the above formulas (20), (21) and (22). These circuits are equivalent to the circuit shown in Fig. 22 and can therefore substitute therefor to achieve the same compression of data.

The formulas (20) to (22) are typical examples of the overall transfer function of the circuit shown in Fig. 22, however it will be apparent that the transfer function can be expressed by various other formulas. Also, the circuits shown in, for example, Figs 13, 15, 18 and 19 can be modified based on the same concept.

A sixth embodiment of the invention will now be described with reference to Figs. 24 to 26.

- 35 The tone signal F1 applied to the input terminal T1 is converted by the ADC 1 into the sampled data F2 which is first compressed by the DPCM 110 to obtain a first compressed data D31. The first compressed data D31 is supplied to a prediction data generating circuit 132. In the prediction data generating circuit 132, the first compressed data D31 is delayed by the serially connected delay circuits 105-1 to 105-p. Outputs of these delay circuits 5-1 to 5-p are multiplied by coefficients b_1 to b_p by multipliers 133-1 to 133-p, respectively. The results of the respective multiplications are added together by the adders 134-1, 134-2, ... to obtain a prediction data FY2 which is supplied to a subtractor 135. The subtractor 135 subtracts the prediction data FY2 from the current difference data D31 to obtain a second compressed data D32. In this case, each of the coefficients b_1 to b_p is smaller than "1" and establishes the following formula (23):

$$45 \quad b_1 + b_2 + \dots + b_p = 1 \quad (23)$$

The prediction data generating circuit 132 and the subtractor 135 constitute a modified PLPC 136. The second compressed data D32 is then supplied to an encoder 138, which may comprises the afore Huffman encoder or the Shannon-Fano encoder, to be further compressed. And an output of the encoder 138 is taken from the output terminal T2 and is written into a memory or transmitted to a remote terminal. In the above circuit, the DPCM 110, modified PLPC 136 and encoder 138 constitute a data compression circuit 100f of this embodiment.

55 The data D31 varies periodically, as shown in Fig. 25, in synchronism with the tone signal F1. And therefore, the value of the current data D31 can be predicted from those data generated "1", "2", ... "p" periods before the current data D31. The prediction data generating circuit 132 is constructed based on the above concept, and can generate a prediction data FY2 very close in value to the current difference data D31 by setting the coefficients b_1 to b_p respectively to proper values. It will be appreciated that the more the delay circuits 105 are provided, the closer to the current data D31 the prediction data FY1 becomes.

And, the closer to the current data D31 the prediction data FY1 is, the higher the efficiency of data compression of the modified PLPC 136 becomes.

A method of determining the coefficients b_1 to b_p will now be described.

The coefficients b_1 to b_p are determined using a statistic method as hereunder described. Assuming
5 that the respective values of the series of data D31 are:

$Y_0, Y_1, Y_2, \dots, Y_m, Y_{m+1}, Y_{m+2}, \dots$

, the modified PLPC 136 independently affects each of m time series, namely,

10

$$Y_0, Y_m, Y_{2m}, \dots \neq Y_{0,j}$$

15

$$Y_1, Y_{m+1}, Y_{2m+1}, \dots \neq Y_{1,j}$$

20

$$Y_{m-1}, Y_{2m-1}, Y_{3m-1}, \dots \neq Y_{m-1,j}$$

25 And therefore, if auto-correlations of $Y_{0,j}$ to $Y_{m-1,j}$ are defined as $r_{0,j}$ to $r_{m-1,j}$, respectively, the coefficients b_1 to b_p can be obtained by solving a normal equation of another auto-correlation which is obtained based on:

30

$$r_j = \sum_{k=0}^{m-1} r_{k,j} \dots \quad (24)$$

More specifically, the input series y_n is first split into the groups shown below:

35

$$Y_{0,j}(j=0, 1, \dots, J) = Y_0, Y_m, Y_{2m}, \dots, Y_{Jm}$$

$$Y_{1,j}(j=0, 1, \dots, J) = Y_1, Y_{m+1}, Y_{2m+1}, \dots, Y_{Jm+1}$$

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$$Y_{m-1,j}(j=0, 1, \dots, J) = Y_{m-1}, Y_{2m-1}, Y_{3m-1}, \dots, Y_{(J+1)m-1}$$

(25)

Then, an auto-correlation is calculated with respect to each of the above formulas as:

50

$$r_{i,n} = \sum_{j=0}^{J-n} r_{i,j} Y_{i,j+n} \dots \quad (26)$$

55

where; $i = 0, 1, \dots, m-1$

And a true auto-correlation $\bar{r}_n$ is obtained by summing the above auto-correlations:

$$\tilde{r}_n = \sum_{i=0}^{m-1} r_{i,n} \quad \dots \quad (27)$$

And therefore, the coefficients b_1 to b_p , which render the data D2 minimum is obtained by solving the next normal equation:

$$\begin{bmatrix} \tilde{r}_0 & \tilde{r}_1 & \dots & \dots & \dots & \tilde{r}_{p-1} \\ \tilde{r}_1 & \tilde{r}_0 & \tilde{r}_1 & & & \cdot \\ \cdot & & & & & \cdot \\ \cdot & & & & & \cdot \\ \cdot & & & & & \cdot \\ \cdot & & & & & \cdot \\ \tilde{r}_{p-1} & \dots & \dots & \dots & \dots & \tilde{r}_0 \end{bmatrix} \begin{bmatrix} b_1 \\ b_2 \\ \cdot \\ \cdot \\ \cdot \\ \cdot \\ b_p \end{bmatrix} = \begin{bmatrix} \tilde{r}_1 \\ \tilde{r}_2 \\ \cdot \\ \cdot \\ \cdot \\ \cdot \\ \tilde{r}_p \end{bmatrix} \quad \dots \quad (28)$$

The above calculation is equivalent to the following procedure:
First, r_n is obtained based on:

$$r_n = \sum_{m=0}^N y_m, y_{m+n} \quad \dots \quad (29)$$

Then, $\tilde{r}_n$ is obtained as:

$$\tilde{r}_n = r_{mn} \quad (30)$$

where; $n = 1, 2, \dots, p$

It is also well known that a linear sum of positive finite auto-correlations is positive finite. And therefore, a demodulation system (see Fig. 26) using the coefficients b_1 to b_p is stable.

Thus, the circuit shown in Fig. 24 can achieve a compression of data more efficiently than those shown in Figs. 13, 18 and 20. Fig. 26 shows a signal reproducing section for reproducing the tone signal F1 from the compressed data D33.

The data D33 applied to the input terminal T3 is decoded by the decoder 139 into a first reproduction data R31 which is supplied to an adder 140. The decoder 139 may be a Huffman decoder or a Shannon-Fano decoder. An output of this adder 140 is supplied to a prediction data generating circuit 141 of the same construction as the prediction data generating circuit 132. And an output of the prediction data generating circuit 141 is supplied to the adder 140 to form a second reproduction data R32. In this case, the adder 140 and the prediction data generating circuit 141 constitute a modified PLPC demodulator 142. The second reproduction data R32 is then converted by the DPCM demodulator 114 into the reproduced sampled data F2 which is supplied to the DAC 10. And the reproduced tone signal F1 is outputted from the DAC 10 through the output terminal T4. The decoder 139, modified PLPC demodulator 142 and DPCM demodulator 114 constitute a data expansion circuit 200f.

A seventh embodiment of the invention will now be described with reference to Figs. 27 and 28.

The tone signal F1 is converted into the sampled data F2 by the ADC 1 and then supplied to serially connected two DPCMs 110 to obtain a first compressed data D41. The first compressed data D41 is supplied to a signal processing circuit 143 which is composed of a plurality of PLPCs 107 connected in series. This signal processing circuit 143 further compresses the first compressed data D41 and outputs a

second compressed data D42 which is supplied to the encoder 138. The encoder converts the second compressed data D42 into a third compressed data D43 and outputs it via the output terminal T2. In this case, the serially connected DPCMs 110, signal processing circuit 143 and encoder 138 constitute a data compression circuit 100g of this seventh embodiment.

5 With this arrangement, if the tone signal F1 varies periodically, the first compressed data D41 also varies periodically. And therefore, the first compressed data D41 can be further compressed by each of the PLPCs 107 of the signal processing circuit 143, whereby the efficiency of data compression of this embodiment is enhanced.

Fig. 28 shows a signal reproducing section of this sixth embodiment. The compressed data D43 fed to 10 the input terminal T3 is decoded into a first reproduction data R41 by a decoder 139 (a Huffman decoder or a Shannon-Fano decoder). The first reproduction data R41 is then supplied to a signal processing circuit 145 which is composed of serially connected PLPC demodulators 117 equal in number to the PLPCs 107 of the signal processing circuit 143 of Fig. 27. The signal processing circuit 145 converts the first reproduction data R41 into a second reproduction data R42. This second reproduction data R42 is then converted by 15 serially connected two DPCM demodulators 114 into the third reproduction data or the sampled data F2 which is converted by the DAC 10 into the reproduced tone signal F1 and supplied to the output terminal T4. In this case, the decoder 139, signal processing circuit 145 and DPCM demodulators 114 constitute a data expansion circuit 200g of this sixth embodiment.

In the above-described embodiment, the DPCMs 110 and the signal processing circuit 143 may 20 alternatively be constructed by analog circuits, in which case the ADC 1 may be disposed downstream of those circuits.

Fig. 29 shows a further modified circuit of the PLPC 107 of the above-described embodiments.

In the case where the input signal F1 is a tone signal, the period of the input signal F1 to be digitally compressed varies under the influence of vibrato and the fluctuation of pitch of the attack portion of the tone 25 signal. In order to improve the efficiency of data compression, the period of the input signal F1 must therefore be detected to control the delay time in the delay circuit 105 of the PLPC 107. The circuit of Fig. 29 is modified in view of this. In Fig. 29, the sampled data F2 is compressed by the LPCC 104 and supplied to a subtractor 150 as well as to the delay circuits 105a, 105 and 105b. The delay circuit 105a delays the output D51 of the LPCC 104 by " $m+1$ " sampling times, and the delay circuit 105b delays the output D51 of 30 the LPCC 104 by " $m-1$ " sampling times. The respective outputs of the delay circuits 105a, 105 and 105b are supplied to a selector 151. The selector 151 selectively outputs to the subtractor 150 one of the outputs of the delay circuits 105a, 105 and 105b in accordance with data L supplied to a selection terminal SE thereof. The sampled data F2 is also supplied to a delay time control circuit 152 which determines the period of the sampled data F2 based in accordance with the variation thereof and outputs the selection data 35 L as the result of the determination. The subtractor 150 subtracts the output of the selector 151 from the output D51 of the LPCC 104 to produce a compressed data D52.

With this arrangement, when the period of the sampled data F2 is equal to " $m+1$ " sampling times, the delay time control circuit 152 outputs the data L of such a value that the selector 151 feeds the output of the delay circuit 105a to the subtractor 150. On the other hand, the selector 151 supplies the output of the 40 delay circuit 105 to the subtractor 150 when the period of the sampled data F2 is equal to " m ", and supplies the output of the delay circuit 105b when the period is equal to " $m-1$ ". Thus, even if the period of the sampled data F2 varies, the subtractor 150 subtracts from the current data that data which was generated exactly one period before. As a result, the efficiency of data compression is further enhanced.

It will be seen that if the data D52 is stored in a memory, the data L need be simultaneously stored in 45 the memory. Also, the delay circuit 105a, 105 and 105b and the selector 151 can be constituted by a RAM or the like. Furthermore, the efficiency of data compression can be further improved by increasing the number of delay circuits 105 interposed between the LPCC 104 and the selector 151 to allow various delay times, namely, " $m-2$ " sampling times, " $m+2$ " sampling times, ... to be selected.

Fig. 30 shows a circuit for reproducing the data F2 from the data D52 compressed by the circuit of Fig. 50 29. The construction and operation of this circuit is self-explanatory, and therefore the description thereof is omitted here.

Claims

- 55 1. A musical tone generating apparatus for an electronic musical instrument comprising means (51, 52) for designating the tone pitch, memory means (M) for storing parameters (D(n)) characterizing a musical waveshape, means (53) for performing a reading operation on the memory means (M) and data reproducing means (200) for computing sample points of the musical waveshape by effecting succes-



second compressed data D42 which is supplied to the encoder 138. The encoder converts the second compressed data D42 into a third compressed data D43 and outputs it via the output terminal T2. In this case, the serially connected DPCMs 110, signal processing circuit 143 and encoder 138 constitute a data compression circuit 100g of this seventh embodiment.

5 With this arrangement, if the tone signal F1 varies periodically, the first compressed data D41 also varies periodically. And therefore, the first compressed data D41 can be further compressed by each of the PLPCs 107 of the signal processing circuit 143, whereby the efficiency of data compression of this embodiment is enhanced.

Fig. 28 shows a signal reproducing section of this sixth embodiment. The compressed data D43 fed to the input terminal T3 is decoded into a first reproduction data R41 by a decoder 139 (a Huffman decoder or a Shannon-Fano decoder). The first reproduction data R41 is then supplied to a signal processing circuit 145 which is composed of serially connected PLPC demodulators 117 equal in number to the PLPCs 107 of the signal processing circuit 143 of Fig. 27. The signal processing circuit 145 converts the first reproduction data R41 into a second reproduction data R42. This second reproduction data R42 is then converted by serially connected two DPCM demodulators 114 into the third reproduction data or the sampled data F2 which is converted by the DAC 10 into the reproduced tone signal F1 and supplied to the output terminal T4. In this case, the decoder 139, signal processing circuit 145 and DPCM demodulators 114 constitute a data expansion circuit 200g of this sixth embodiment.

10 In the above-described embodiment, the DPCMs 110 and the signal processing circuit 143 may alternatively be constructed by analog circuits, in which case the ADC 1 may be disposed downstream of those circuits.

Fig. 29 shows a further modified circuit of the PLPC 107 of the above-described embodiments.

In the case where the input signal F1 is a tone signal, the period of the input signal F1 to be digitally compressed varies under the influence of vibrato and the fluctuation of pitch of the attack portion of the tone signal. In order to improve the efficiency of data compression, the period of the input signal F1 must therefore be detected to control the delay time in the delay circuit 105 of the PLPC 107. The circuit of Fig. 29 is modified in view of this. In Fig. 29, the sampled data F2 is compressed by the LPCC 104 and supplied to a subtractor 150 as well as to the delay circuits 105a, 105 and 105b. The delay circuit 105a delays the output D51 of the LPCC 104 by "m + 1" sampling times, and the delay circuit 105b delays the output D51 of the LPCC 104 by "m - 1" sampling times. The respective outputs of the delay circuits 105a, 105 and 105b are supplied to a selector 151. The selector 151 selectively outputs to the subtractor 150 one of the outputs of the delay circuits 105a, 105 and 105b in accordance with data L supplied to a selection terminal SE thereof. The sampled data F2 is also supplied to a delay time control circuit 152 which determines the period of the sampled data F2 based in accordance with the variation thereof and outputs the selection data L as the result of the determination. The subtractor 150 subtracts the output of the selector 151 from the output D51 of the LPCC 104 to produce a compressed data D52.

With this arrangement, when the period of the sampled data F2 is equal to "m + 1" sampling times, the delay time control circuit 152 outputs the data L of such a value that the selector 151 feeds the output of the delay circuit 105a to the subtractor 150. On the other hand, the selector 151 supplies the output of the delay circuit 105 to the subtractor 150 when the period of the sampled data F2 is equal to "m", and supplies the output of the delay circuit 105b when the period is equal to "m - 1". Thus, even if the period of the sampled data F2 varies, the subtractor 150 subtracts from the current data that data which was generated exactly one period before. As a result, the efficiency of data compression is further enhanced.

It will be seen that if the data D52 is stored in a memory, the data L need be simultaneously stored in the memory. Also, the delay circuit 105a, 105 and 105b and the selector 151 can be constituted by a RAM or the like. Furthermore, the efficiency of data compression can be further improved by increasing the number of delay circuits 105 interposed between the LPCC 104 and the selector 151 to allow various delay times, namely, "m - 2" sampling times, "m + 2" sampling times, ... to be selected.

Fig. 30 shows a circuit for reproducing the data F2 from the data D52 compressed by the circuit of Fig. 29. The construction and operation of this circuit is self-explanatory, and therefore the description thereof is omitted here.

Claims

1. A musical tone generating apparatus for an electronic musical instrument comprising means (51, 52) for designating the tone pitch, memory means (M) for storing parameters (D(n)) characterizing a musical waveshape, means (53) for performing a reading operation on the memory means (M) and data reproducing means (200) for computing sample points of the musical waveshape by effecting succes-



- sively linear operations on a series of already computed sample points on the basis of said stored parameters, characterized in that said reading operation is performed at predetermined time intervals corresponding to the tone pitch, the parameters stored in the memory means (M) are a series of difference data (D(n)) each representing a difference between an individual sample data (F(n)) of an original musical waveshape and an individual prediction data (FY(n)), the individual prediction data (FY(n)) having been determined by effecting a first linear operation on some of the sample data (F(n)) of the original musical waveshape which have a predetermined relation of time to the individual sample data (F(n)) for which the individual prediction data (FY(n)) is being determined and the data reproducing means (200) include means for adding to each individual of said difference data (D(n)) a further individual data obtained by effecting a second linear operation corresponding to said first linear operation on some of the already computed sample points which have said predetermined relation of time to said outputted individual difference data (D(n)).
2. A musical tone generating apparatus according to claim 1, wherein said memory means (M) stores a series of second difference data (En) each representing a difference between a corresponding one of said series of difference data and that of the series of difference data produced a time interval corresponding to a period of said tone signal before said corresponding one of said series of difference data, said musical tone generating apparatus further comprising
second data reproducing means which comprises first delay circuit means (32) for delaying data inputted thereto by said time interval to output delayed data, and first adder means (31) for adding said each second difference data outputted from said memory means (M) to said delayed data, each addition result being supplied to said first delay circuit means as said data and to said data reproducing means (200) as said outputted difference data.
 3. A musical tone generating apparatus according to claim 2, wherein said memory means (M) further stores variation data representative of variation of period of said musical tone signal together with said second difference data, said first delay circuit means varying said time interval in accordance with said variation data outputted from said memory means (M).
 4. A musical tone generating apparatus according to claim 2, wherein said first delay circuit means further comprises multiplier means (40-44) for multiplying one of input and output data thereof by a predetermined coefficient.
 5. A musical tone generating apparatus according to claim 2, wherein said second data reproducing means comprising a plurality of circuit means (36-39) connected in series and each having the same construction as said second data reproducing means.
 6. A musical tone generating apparatus according to claim 2, wherein said memory means (M) stores a series of third difference data each representing a difference between a corresponding one of said series of second difference data (En) and that of the series of second difference data produced a second time interval corresponding to a predetermined number of sampling intervals of said tone signal before said corresponding one of said series of second difference data, said musical tone generating apparatus further comprising
third data reproducing means comprising second delay circuit means for delaying data inputted thereto by said second time interval to output delayed data, and second adder means for adding each of said third difference data outputted from said memory means to said delayed data outputted from said second delay circuit means, each addition result being supplied to said second delay circuit means as said data and to said first adder means as said outputted second difference data.
 7. A musical tone generating apparatus according to claim 6, wherein said third data reproducing means comprising a plurality of circuit means connected in series and each having the same construction as said third data reproducing means.
 8. A musical tone generating apparatus according to any one of claims 1 to 7, wherein said each prediction data (FY(n)) is obtained by effecting said first linear operation on those of said series of sample data produced before and after said individual sample data (F(n)), said those of said sample data including at least three of said series of sample data.



9. A musical tone generating apparatus according to claim 8, wherein said prediction data (FY1) is obtained through:
- a plurality of stages of third delay circuit means (110) whose first stage is supplied with said individual sample data, each of said stages delaying an input thereto by a third time interval corresponding to said sampling interval to produce a delayed output;
 - a plurality of second multiplier means (111) each multiplying said delayed output from a respective one of said stages by a predetermined coefficient to output a multiplication result; and
 - third adder means (113) for adding all multiplication results of said plurality of second multiplier means (111) together to form said prediction data (FY1).
10. A musical tone generating apparatus according to claim 9, wherein said data reproducing means comprises:
- fourth adder means (119);
 - a plurality of stages of fourth delay circuit means (110a) equal in number of stage to said third delay circuit means (110), the first stage of said fourth delay circuit means (110a) being supplied with an output of said fourth adder means (119), each of said stages delaying an input thereto by said third time interval to produce a delayed output;
 - a plurality of third multiplier means (111a) each multiplying said delayed output from a respective one of said stages of said fourth delay circuit means (110a) by said predetermined coefficient to output a multiplication result; and
 - fifth adder means (113a) adding all multiplication results of said plurality of third multiplier means together to output an addition result;
 - said fourth adder means (119) adding said addition result to said each outputted difference data to form said corresponding one of said sample data.
11. A musical tone generating apparatus according to claim 8, wherein said prediction data is obtained through:
- a plurality of stages of fifth delay circuit means whose first stage is supplied with said individual sample data, each of said stages delaying an input thereto by a fourth time interval corresponding to a predetermined number of periods of said tone signal to produce a delayed output;
 - a plurality of fourth multiplier means each multiplying said delayed output from a respective one of said stages by a predetermined coefficient to output a multiplication result; and
 - sixth adder means for adding all multiplication results of said plurality of fourth multiplier means together to form said prediction data.
12. A musical tone generating apparatus according to claim 11, wherein said data reproducing means comprises:
- seventh adder means;
 - a plurality of stages of sixth delay circuit means equal in number of stage to said fifth delay circuit means (110), the first stage of said sixth delay circuit means being supplied with an output of said seventh adder means, each of said stages delaying an input thereto by said fourth time interval to produce a delayed output;
 - a plurality of fifth multiplier means each multiplying said delayed output from a respective one of said stages of said sixth delay circuit means by said predetermined coefficient to output a multiplication result; and
 - eighth adder means adding all multiplication results of said plurality of fifth multiplier means together to output an addition result;
 - said seventh adder means adding said addition result from said eighth adder means to said each outputted difference data to form said corresponding one of said sample data.
13. A musical tone data compression apparatus for a musical tone generating apparatus of an electronic musical instrument comprising:
- sampling means (ADC) for sampling plural amplitude values corresponding to plural points on the time axis of a musical tone signal $F(t)$ to be produced; and
 - prediction means for calculating prediction amplitude values corresponding to said plural points by effecting a linear operation on said amplitude values, each of said prediction amplitude values being determined on the basis of a point or points before and after the point corresponding to each of said amplitude values,



said musical tone data compression apparatus further providing compressing means for sequentially compressing said amplitude values based on said prediction amplitude values to thereby produce compressed amplitude values which are to be stored in a memory.

- 5 14. A musical tone data compression apparatus according to claim 13 wherein said compressing means comprises first subtracting means (3) which subtracts said prediction amplitude values from said amplitude values respectively and thereby outputs a first subtracted result ($D(n-1)$).
- 10 15. A musical tone data compression apparatus according to claim 14 wherein said compressing means further comprises
- delay means for delaying said first subtracted result by a period and
- second subtracting means (23) which subtracts delayed first subtracted result from said first subtracted result and thereby outputs a second subtracted result.

16 Revendications

1. Appareil de production d'une tonalité musicale pour un instrument de musique électronique comprenant un moyen (51, 52) pour indiquer la hauteur de son, un moyen de mémoire (M) pour mémoriser des paramètres ($D(n)$) caractérisant une forme d'onde musicale, un moyen (53) pour effectuer une opération de lecture dans le moyen de mémoire (M) et un moyen de reproduction (200) pour calculer des points échantillons de la forme d'onde musicale en effectuant de manière successive des opérations linéaires sur une série de points échantillons déjà calculés sur la base desdits paramètres mémorisés, caractérisé en ce que ladite opération de lecture est effectuée à des intervalles de temps prédéterminés correspondant à la hauteur de son, en ce que les paramètres mémorisés dans le moyen de mémoire (M) sont une série de données de différence ($D(n)$) représentant chacun une différence entre une donnée échantillon individuelle ($F(n)$) d'une forme d'onde musicale d'origine et une donnée prévue individuelle ($FY(n)$), la donnée prévue individuelle ($FY(n)$) ayant été déterminée en effectuant une première opération linéaire sur certaines des données échantillons ($F(n)$) de l'onde de forme musicale d'origine qui avaient une relation prédéterminée de temps avec la donnée échantillon individuelle ($F(n)$) pour laquelle la donnée prévue individuelle ($FY(n)$) est en cours de détermination et en ce que le moyen de reproduction de données (200) comprend un moyen pour additionner à chacune individuelle desdites données de différence ($D(n)$) une donnée individuelle supplémentaire obtenue en effectuant une seconde opération linéaire correspondant à ladite première opération linéaire sur certains des points échantillons déjà calculés qui ont ladite relation de temps prédéterminée avec ladite donnée de différence individuelle sortie ($D(n)$).
2. Appareil de production d'une tonalité musicale selon la revendication 1, dans lequel ledit moyen de mémoire (M) mémorise une série de secondes données de différence ($E(n)$) représentant chacune une différence entre l'une correspondante de ladite série de données de différence et celle de la série de données de différence produite à un intervalle de temps correspondant à une période dudit signal de tonalité avant ladite une correspondante de ladite série de données de différence, ledit appareil de production de tonalité musicale comprenant de plus
- un second moyen de reproduction de données qui comprend un premier moyen de circuit à retard (32) pour retarder la donnée qui y est entrée dudit intervalle de temps pour sortir la donnée retardée, et un premier moyen additionneur (31) pour additionneur chacune desdites secondes données de différence sorties dudit moyen de mémoire (M) à ladite donnée retardée, chaque résultat d'addition étant délivré audit premier moyen de circuit à retard en tant que ladite donnée et audit moyen de reproduction de données (200) en tant que ladite donnée de différence sortie.
3. Appareil de production de tonalité musicale selon la revendication 2, dans lequel ledit moyen de mémoire (M) mémorise en outre une donnée de variation représentative de la variation de période dudit signal de tonalité musicale en même temps que ladite seconde donnée de différence, ledit premier moyen de circuit à retard modifiant ledit intervalle de temps en fonction de ladite donnée de variation sortie dudit moyen de mémoire (M).
4. Appareil de production de tonalité musicale selon la revendication 2, dans lequel ledit premier moyen de circuit à retard comprend de plus un moyen multiplieur (40 à 44) pour multiplier l'une des données d'entrée et de sortie de celui-ci par un coefficient prédéterminé.



5. Appareil de production de tonalité musicale selon la revendication 2, dans lequel ledit second moyen de reproduction de données comprend une pluralité de moyens de circuit (36 à 39) connectés en série et comportant chacun la même structure que celle dudit second moyen de reproduction de données.
- 5 6. Appareil de production de tonalité musicale selon la revendication 2, dans lequel ledit moyen de mémoire (M) mémorise une série de troisièmes données de différence chacune représentant une différence entre l'une correspondante de ladite série de secondes données de différence (En) et celle de la série de secondes données de différence produite à un intervalle de temps correspondant à un nombre prédéterminé d'intervalles d'échantillonnage dudit signal de tonalité avant ladite une correspon-
- 10 dante de ladite série de secondes données de différence, ledit appareil de production de tonalité musicale comprenant de plus
un troisième moyen de reproduction de données comprenant un second moyen de circuit à retard pour retarder la donnée qui y est entrée dudit second intervalle de temps pour sortir la donnée retardée, et un second moyen additionneur pour additionner chacune desdites troisièmes données de
- 15 différence sorties dudit moyen de mémoire à ladite donnée retardée sortie dudit second moyen de circuit à retard, chaque résultat d'addition étant délivré audit second moyen de circuit à retard en tant que ladite donnée et audit premier moyen additionneur en tant que ladite seconde donnée de différence sortie.
- 20 7. Appareil de production de tonalité musicale selon la revendication 6, dans lequel ledit troisième moyen de reproduction de données comprend une pluralité de moyens de circuit connectés en série et ayant chacun la même structure que celle dudit troisième moyen de reproduction de données.
8. Appareil de production de tonalité musicale selon l'une quelconque des revendications 1 à 7, dans
- 25 lequel chacune desdites données prévues (FY(n)) est obtenue en effectuant ladite première opération linéaire sur celles de ladite série de données échantillons produites avant et après ladite donnée échantillon individuelle (F(n)), celles dites desdites données échantillons comprenant au moins trois données échantillons de ladite série de données échantillons.
- 30 9. Appareil de production de tonalité musicale selon la revendication 8, dans lequel ladite donnée prévue (FY1) est obtenue au moyen :
d'une pluralité d'étages dudit troisième moyen de circuit à retard (110) dont le premier étage est alimenté avec ladite donnée échantillon individuelle, chacun desdits étages retardant une entrée dans celui-ci d'un troisième intervalle de temps correspondant audit intervalle d'échantillonnage pour produ-
- 35 re une sortie retardée;
une pluralité de seconds moyens multiplieurs (111) multipliant chacun ladite sortie retardée de l'un respectif desdits étages par un coefficient prédéterminé pour sortir un résultat de multiplication; et
un troisième moyen additionneur (113) pour additionner tous les résultats de multiplication de ladite pluralité de seconds moyens multiplieurs (111) ensemble pour former ladite donnée prévue (FY1).
- 40 10. Appareil de production de tonalité musicale selon la revendication 9, dans lequel ledit moyen de reproduction de données comprend :
un quatrième moyen additionneur (119);
une pluralité d'étages du quatrième moyen de circuit à retard (110a) égaux en nombre d'étages
- 45 audit troisième moyen de circuit à retard (110), le premier étage dudit quatrième moyen de circuit à retard (110a) étant alimenté avec une sortie dudit quatrième moyen additionneur (119), chacun desdits étages retardant une entrée de celui-ci dudit troisième intervalle de temps pour produire une sortie retardée;
une pluralité de troisièmes moyens multiplieurs (111a) multipliant chacun ladite sortie retardée de l'un respectif desdits étages dudit quatrième moyen de circuit à retard (110a) par ledit coefficient
- 50 prédéterminé pour sortir un résultat de multiplication; et
un cinquième moyen additionneur (113a) additionnant tous les résultats de multiplication de ladite pluralité de troisièmes moyens multiplieurs ensemble pour sortir un résultat d'addition;
ledit quatrième moyen additionneur (119) additionnant ledit résultat d'addition à chacune desdites
- 55 données de différence sorties pour former ladite une correspondante desdites données échantillons.
11. Appareil de production de tonalité musicale selon la revendication 8, dans lequel ladite donnée prévue est obtenue au moyen :



d'une pluralité d'étages d'un cinquième moyen de circuit à retard dont le premier étage est alimenté avec ladite donnée échantillon individuelle, chacun desdits étages retardant une entrée dans celui-ci d'un quatrième intervalle de temps correspondant à un nombre prédéterminé de périodes dudit signal de tonalité pour produire une sortie retardée;

- 5 une pluralité de quatrièmes moyens multiplieurs multipliant chacun ladite sortie retardée à partir de l'un respectif desdits étages par un coefficient prédéterminé pour sortir un résultat de multiplication; et un sixième moyen additionneur pour additionner tous les résultats de multiplication de ladite pluralité de quatrièmes moyens multiplieurs ensemble pour former ladite donnée prévue.

- 10 12. Appareil de production de tonalité musicale selon la revendication 11, dans lequel ledit moyen de reproduction de données comprend :

un septième moyen additionneur;

- 15 une pluralité d'étages du sixième moyen de circuit à retard égaux en nombre d'étages audit cinquième moyen de circuit à retard (110), le premier étage dudit sixième moyen de circuit à retard étant alimenté avec une sortie dudit septième moyen additionneur, chacun desdits étages retardant une entrée de celui-ci dudit quatrième intervalle de temps pour produire une sortie retardée;

une pluralité de cinquièmes moyens multiplieurs multipliant chacun ladite sortie retardée de l'un respectif desdits étages dudit sixième moyen de circuit à retard par ledit coefficient prédéterminé pour sortir un résultat de multiplication; et

- 20 un huitième moyen additionneur additionnant tous les résultats de multiplication de ladite pluralité de cinquièmes moyens multiplieurs ensemble pour sortir un résultat d'addition;

ledit septième moyen additionneur additionnant ledit résultat d'addition dudit huitième moyen additionneur à chacune desdites données de différence sorties pour former ladite une correspondante desdites données échantillons.

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13. Appareil de compression de données de tonalité musicale pour un appareil de production de tonalité musicale d'un instrument de musique électronique comprenant:

un moyen d'échantillonnage (ADC) pour échantillonner plusieurs valeurs d'amplitude correspondant à plusieurs points sur l'axe des temps d'un signal de tonalité musicale $F(t)$ devant être produit; et

- 30 un moyen de prévision pour calculer des valeurs d'amplitude prévues correspondant auxdits plusieurs points en effectuant une opération linéaire sur lesdites valeurs d'amplitude, chacune desdites valeurs d'amplitude prévues étant déterminées sur la base d'un point ou de points avant et après le point correspondant à chacune desdites valeurs d'amplitude,

- 35 ledit appareil de compression de données de tonalité musicale procurant en outre des moyens de compression pour comprimer de manière séquentielle lesdites valeurs d'amplitude en se basant sur lesdites valeurs d'amplitude prévues pour produire par ce moyen des valeurs d'amplitude comprimées qui doivent être mémorisées dans une mémoire.

- 40 14. Appareil de compression de données de tonalité musicale selon la revendication 13 dans lequel ledit moyen de compression comprend un premier moyen de soustraction (3) qui soustrait lesdites valeurs d'amplitude prévues desdites valeurs d'amplitude respectivement et qui sort par ce moyen un premier résultat soustrait ($D(n-1)$).

- 45 15. Appareil de compression de données de tonalité musicale selon la revendication 14 dans lequel ledit moyen de compression comprend en outre

un moyen de retard pour retarder ledit premier résultat soustrait d'une période; et

un second moyen de soustraction (23) qui soustrait le premier résultat soustrait retardé dudit premier résultat soustrait et sort par ce moyen un second résultat soustrait.

50 Patentansprüche

1. Musiktongenerator für ein elektronisches Musikinstrument mit Mitteln (51, 52) zur Bestimmung der Tonhöhe, mit Speichermitteln (M) zur Speicherung von Parametern ($D(n)$), die eine Musiktonwellenform charakterisieren, mit Mitteln (53) zur Durchführung einer Leseoperation in den Speichermitteln (M) und mit Datenreproduzierungsmitteln (200) zur Berechnung von Abtastwerten der Musiktonwellenform, wobei sukzessive lineare Operationen auf eine Reihe von bereits berechneten Abtastwerten auf der Basis der gespeicherten Parameter durchgeführt werden, dadurch gekennzeichnet, daß die Leseoperation zu vorgegebenen, der Tonhöhe entsprechenden Zeitintervallen durchgeführt wird, und daß die in den



Speichermitteln (M) gespeicherten Parameter eine Serie von Differenzdaten (D(n)) sind, die jeweils eine Differenz zwischen einem individuellen Abtastdatenwert (F(n)) einer Originalmusiktonwellenform und einem individuellen Vorhersagedatenwert (FY(n)) sind, wobei der individuelle Vorhersagedatenwert (FY(n)) dadurch bestimmt worden ist, daß eine erste lineare Operation auf einige der Abtastdaten (F(n)) der Originalmusiktonwellenform durchgeführt worden ist, die eine vorgegebene Zeitbeziehung zu dem individuellen Abtastdatenwert (F(n)) haben, für die der individuelle Vorhersagedatenwert (FY(n)) bestimmt wird, und daß die Datenreproduzierungsmittel (200) Mittel aufweisen, um zu jedem individuellen Wert der Differenzdaten (D(n)) einen weiteren individuellen Datenwert zu addieren, der dadurch erhalten wird, daß eine zweite lineare Operation, die der ersten linearen Operation entspricht, auf einige der bereits berechneten Abtastpunkte durchgeführt wird, die die genannte vorgegebene Zeitbeziehung zu dem ausgegebenen individuellen Differenzdatenwert (D(n)) aufweisen.

2. Musikongenerator nach Anspruch 1, bei dem die Speichermittel (M) eine Reihe von zweiten Differenzdaten (En) speichern, die jeweils eine Differenz zwischen einem entsprechenden Datenwert aus der Serie der Differenzdaten und einem solchen aus der Serie der Differenzdaten darstellen, die ein der Periode des Tonsignales entsprechendes Zeitintervall vor der entsprechenden Serie von Differenzdaten erzeugt worden ist,

welcher Musikongenerator des weiteren zweite Datenreproduzierungsmittel aufweist, die erste Verzögerungsschaltungsmittel (32) umfassen, um an sie eingegebene Daten um das genannte Zeitintervall zu verzögern und verzögerte Daten auszugeben, sowie erste Addiermittel (31), um jeweils den genannten zweiten Differenzdatenwert, der von den Speichermitteln (M) ausgegeben worden ist, zu den verzögerten Daten zu addieren, wobei jedes Additionsergebnis den ersten Verzögerungsschaltungsmitteln als diese Daten und den Datenreproduzierungsmitteln (200) als die ausgegebenen Differenzdaten zugeführt werden.

3. Musikongenerator nach Anspruch 2, bei dem die Speichermittel (M) des weiteren für eine Variation der Periode des Musiktonsignales repräsentative Variationsdaten zusammen mit den zweiten Differenzdaten speichern, wobei die ersten Verzögerungsschaltungsmittel das Zeitintervall entsprechend den von den Speichermitteln (M) ausgegebenen Variationsdaten variieren.

4. Musikongenerator nach Anspruch 2, bei dem die ersten Verzögerungsschaltungsmittel des weiteren Multipliziermittel (40-44) aufweisen, um die Eingangs- oder Ausgangsdaten mit einem vorgegebenen Koeffizienten zu multiplizieren.

5. Musikongenerator nach Anspruch 2, bei dem die zweiten Datenreproduzierungsmittel eine Vielzahl von Schaltungsmitteln (36-39) aufweisen, die in Serie geschaltet sind und die jeweils den gleichen Aufbau wie die zweiten Datenreproduzierungsmittel aufweisen.

6. Musikongenerator nach Anspruch 2, bei dem die Speichermittel (M) eine Serie von dritten Differenzdaten speichern, die jeweils eine Differenz zwischen einem entsprechenden Datenwert aus der Serie der zweiten Differenzdaten (En) und einem solchen aus der Serie der zweiten Differenzdaten darstellen, die um ein zweites einer vorgegebenen Anzahl von Abtastintervallen des Tonsignales entsprechendes Zeitintervall vor dem entsprechenden Datenwert aus der Serie der zweiten Differenzdaten erzeugt werden, welcher Musikongenerator des weiteren folgendes aufweist:

dritte Datenreproduzierungsmittel, die zweite Verzögerungsschaltungsmittel aufweisen, um an sie eingegebene Daten um das zweite Intervall zu verzögern und verzögerte Daten auszugeben, sowie zweite Addiermittel, um jedes der dritten Differenzdaten, die von den Speichermitteln ausgegeben werden, zu den verzögerten Daten, die von den zweiten Verzögerungsschaltungsmitteln ausgegeben werden, zu addieren, wobei jedes Additionsergebnis den zweiten Verzögerungsschaltungsmitteln als diese Daten und den ersten Addiermitteln als diese ausgegebenen zweiten Differenzdaten zugeführt werden.

7. Musikongenerator nach Anspruch 6, bei dem die dritten Datenreproduzierungsmittel eine Vielzahl von Schaltungsmitteln aufweisen, die in Serie geschaltet sind und jeweils den gleichen Aufbau wie die dritten Datenreproduzierungsmittel aufweisen.

8. Musikongenerator nach einem der Ansprüche 1 bis 7, bei dem jeder Vorhersagedatenwert (FY(n)) durch Durchführung der ersten linearen Operation auf diejenigen aus der Serie der Abtastdaten gewonnen wird, die vor und nach dem individuellen Abtastdatenwert (F(n)) erzeugt werden, wobei diese



Daten der Abtastdaten mindestens drei aus der Serie der Abtastdaten einschließen.

9. Musikongenerator nach Anspruch 8, bei dem der Vorhersagedatenwert (FY1) durch folgende Mittel erhalten wird:
 - 5 eine Vielzahl von Stufen von dritten Verzögerungsschaltungsmitteln (110), deren erste Stufe mit dem individuellen Abtastdatenwert beaufschlagt wird, wobei jede der Stufe ein an sie eingegebenes Eingangssignal um ein drittes Zeitintervall verzögert, das dem Abtastintervall entspricht, um ein verzögertes Ausgangssignal zu erzeugen;
 - eine Vielzahl von zweiten Multipliziermitteln (111), von denen jedes das verzögerte Ausgangssignal
 - 10 aus den jeweiligen Stufen mit einem vorgegebenen Koeffizienten multipliziert, um ein Multiplikationsergebnis auszugeben und
 - dritte Addiermittel (113), um alle Multiplikationsergebnisse aus der Vielzahl der zweiten Multipliziermittel (111) zusammen zu addieren, um den Vorhersagewert (FY1) zu bilden.
- 15 10. Musikongenerator nach Anspruch 9, bei dem die Datenreproduzierungsmittel folgendes aufweisen:
 - vierte Addiermittel (119);
 - eine Vielzahl von Stufen von vierten Verzögerungsschaltungsmitteln (110a), deren Anzahl gleich
 - der Anzahl der Stufen der dritten Verzögerungsschaltungsmittel (110) ist, wobei die erste Stufe der
 - vierten Verzögerungsschaltungsmittel (110a) mit einem Ausgangssignal der vierten Addiermittel (119)
 - 20 beaufschlagt wird, und jede der Stufen einen an sie eingegebenen Eingangssignal um das dritte Zeitintervall verzögert und ein verzögertes Ausgangssignal liefert;
 - eine Vielzahl von dritten Multipliziermitteln (111 a), von denen jedes das verzögerte Ausgangssignal
 - aus der jeweiligen Stufe der vierten Verzögerungsschaltungsmittel (110a) mit dem vorgegebenen Koeffizienten multipliziert, um ein Multiplikationsergebnis auszugeben; und
 - 25 fünfte Addiermittel (113a), die alle Multiplikationsergebnisse aus der Vielzahl der dritten Multipliziermittel zusammenzählen, um ein Additionsergebnis auszugeben;
 - wobei die vierten Addiermittel (119) das Additionsergebnis zu dem jeweiligen ausgegebenen Differenzdatenwert hinzuaddieren, um den entsprechenden Abtastdatenwert zu bilden.
- 30 11. Musikongenerator nach Anspruch 8, bei dem der Vorhersagedatenwert mit folgenden Mitteln erhalten wird:
 - eine Vielzahl von Stufen von fünften Verzögerungsmitteln, deren erste Stufe mit dem individuellen
 - Abtastwert beaufschlagt wird und deren jede Stufe einen an diese eingegebenen Eingangssignalwert
 - um ein viertes Zeitintervall verzögert, das einer vorgegebenen Anzahl von Perioden des Musiktonsigna-
 - 35 les entspricht, wobei ein verzögertes Ausgangssignal erzeugt wird;
 - eine Vielzahl von vierten Multipliziermitteln, von denen jedes den verzögerten Ausgangswert
 - von der jeweiligen Stufe mit einem vorgegebenen Koeffizienten multipliziert, um ein Multiplikationser-
 - gebnis auszugeben und
 - sechste Addiermittel, um alle Multiplikationsergebnisse der Vielzahl von vierten Multipliziermittel
 - 40 zusammen zu addieren und so den Vorhersagedatenwert zu bilden.
12. Musikongenerator nach Anspruch 11, bei dem die Datenreproduzierungsmittel folgendes aufweisen:
 - siebte Addiermittel;
 - eine Vielzahl von Stufen von sechsten Verzögerungsschaltungsmitteln, deren Anzahl gleich der
 - 45 Anzahl der Stufen der fünften Verzögerungsschaltungsmittel (110) ist, wobei die erste Stufe der
 - sechsten Verzögerungsschaltungsmittel mit einem Ausgangssignal der siebten Addiermittel beaufschlagt
 - wird, und jede der Stufen ein an sie gegebenes Eingangssignal um das vierte Zeitintervall verzögert,
 - und so ein verzögertes Ausgangssignal bildet;
 - eine Vielzahl von fünften Multipliziermitteln, von denen jedes den verzögerten Ausgangswert aus
 - 50 einer jeweiligen Stufe der sechsten Verzögerungsschaltungsmittel mit dem vorgegebenen Koeffizienten
 - multipliziert, und so ein Multiplikationsergebnis ausgibt; und
 - achte Addiermittel, die alle Multiplikationsergebnisse aus der Vielzahl der fünften Multipliziermittel
 - zusammenaddieren, um ein Additionsergebnis auszugeben; und
 - siebte Addiermittel, die das Additionsergebnis der achten Addiermittel zu dem jeweiligen ausgege-
 - 55 benen Differenzdatenwert hinzuaddieren, um den entsprechenden Wert der Abtastwerte zu bilden.
13. Einrichtung zur Musiktondatenkompression für einen Musikongenerator eines elektronischen Musikin-
- strumentes, der folgendes aufweist:



Abtastmittel (ADC) zum Abtasten von mehreren Amplitudenwerten, die einer Vielzahl von Punkten auf der Zeitachse eines zu erzeugenden Musiksignals ($F(t)$) entsprechen; und

Vorhersagemittel zur Berechnung von den verschiedenen Punkten entsprechenden Vorhersageamplitudenwerten durch Ausübung einer linearen Operation auf die Amplitudenwerte, wobei jeder der Vorhersageamplitudenwerte auf der Grundlage eines Punktes oder einiger Punkte vor und nach dem dem jeweiligen Amplitudenwert entsprechenden Punkt bestimmt wird,

welche Einrichtung zur Musikdatenkompression des weiteren Kompressionsmittel aufweist, um sequenziell die Amplitudenwerte auf der Grundlage der Vorhersageamplitudenwerte zu komprimieren, um dadurch komprimierte Amplitudenwerte zu erzeugen, die in einem Speicher zu speichern sind.

14. Einrichtung zur Musikdatenkompression nach Anspruch 13, bei dem die Kompressionsmittel erste Subtraktionsmittel (3) aufweisen, die die Vorhersageamplitudenwerte von den jeweiligen Amplitudenwerten abziehen und ein erstes Subtraktionsergebnis ($D(n-1)$) ausgeben.

15. Einrichtung zur Musikdatenkompression nach Anspruch 14, bei der die Kompressionsmittel des weiteren folgendes aufweisen:

Verzögerungsmittel zur Verzögerung des ersten Subtraktionsergebnisses um eine Periode und zweite Subtraktionsmittel (23), die das verzögerte erste Subtraktionsergebnis von dem ersten Subtraktionsergebnis subtrahieren und ein zweites Subtraktionsergebnis ausgeben.