Description

The invention relates to a method of correcting erroneous values of samples of an equidistantly sampled signal by means of a recursion formula expressing the erroneous samples as a weighted sum of the values of a number of preceding samples. The invention also relates to a device for carrying out the method.

Such a method may be applied to pulse-code modulated systems as used for the transmission and processing of audio and speech signals. The method is particularly suitable for use in the "Compact Disc Digital Audio System" as described in Philips Technical Review, 40, 1982, No. 9, 1981/1982. In this system samples of an analogue audio signal are recorded on a disc ("Compact Disc") in digitally coded form. Flaws or fingermarks etc. on the disc may give rise to errors in reading the digital information. In order to corrent these errors the digital information is recorded on the disc in scrambled form and in addition error-correction codes are recorded on the disc. This enables some errors to be corrected completely. However, in the case of scratches etc. a multitude of consecutive errors, a so-called burst, may occur. If the number of errors is larger than can be corrected by means of the error correction code, these errors are merely detected. Due to these errors one or more samples with an incorrect value may be detected during read-out after the information has been descrambled. If one incorrect sample is detected its value is estimated by linear interpolation between the values of the preceding and the following sample. Alternatively, the value of the incorrect sample may be selected to be equal to that of the preceding sample. If two or more consecutive incorrect samples are detected, the values of these samples are made equal to zero ("muting"). In order to obtain a gradual transition the values of some of the preceding and following samples change gradually to zero.

European Patent Application 44963 discloses an interpolation method for correcting a maximum of four consecutive incorrect samples. In accordance with this method the value of each incorrect sample is estimated by taking a weighted means of the value of the next correct sample and the value of the preceding correct sample or the preceding estimate, the weighting factors of the weighted mean being determined by the number of missing samples. In the case of more than four consecutive incorrect samples the value of the last correct sample is assigned to such a number of samples that four incorrect samples are left. The values of these four incorrect samples are then estimated by means of the said interpolation method.

Apart from the fact that this known interpolation method is comparatively inaccurate, it is not suitable for the correction of more than four consecutive incorrect samples.

A method of the type described in the opening paragraph is known from JP—A—5342714. When an erroneous values is detected the sample is replaced by a sample having a value equal to the weighted sum of a number of preceding samples. Since the weighting factors have fixed values and only a number of preceding samples equal to the number of terms of the recursion formula is used to obtain a corrected value of a sample, the known method is rather inaccurate.

Therefore, it is an object of the invention to provide an interpolation method which yields more accurate results and which enables a large number of errors to be corrected in an accurate manner. According to the invention a method of a type specified in the opening sentence is characterized in that the steps of

— deriving from the number of samples having erroneous values a first sampling interval within which the samples having erroneous values are situated,

— deriving from the number of samples having erroneous values the number of samples whose values are weighted in the recursion formula,

— determining the weighting factors of the recursion formula from the recursion formulas with unknown weighting factors for a number of samples in the first sampling interval using the values of the samples in that interval, a first estimate being taken for the values of the samples having erroneous values,

— determining from the number of samples whose values are weighted in the recursion formula a second sampling interval within the first sampling interval and within which the samples having erroneous values are situated,

— deriving estimated values for the samples having erroneous values from the recursion formulas with known weighting factors for a number of samples in the second interval using the values of the correct samples in that interval.

The invention is based on the recognition of the fact that for signals whose spectral characteristics do not vary too rapidly in time, such as audio and speech signals, the signal in a specific interval can be defined accurately by a best-fitting recursion formula with a finite number of terms. This recursion formula is determined by means of the values of the samples in said interval.

Subsequently, the values of the incorrect samples to be corrected are selected to be in conformity with the best-fitting recursion formula thus found. It is to be noted that describing values of samples by means of best-fitting recursion formulae is known *per se* from the article "Linear Prediction: A Tutorial Review" Proc. of the IEEE, Vol. 63, No. 4, April 1975; 561—580. However, in this article the recursion is not employed for computing the values of incorrect samples but for reducing the amount of data to be transmitted in transmission systems.

In accordance with the invention the following steps are carried out for determining the weighting factors of the terms of the recursion formula:

— drawing up a recursion formula for a number of samples within the interval, which formula contains a weighted sum of the values of a number of preceding samples, the difference between the value of a sample and the value of the weighted sum being expressed as an error term, a first estimate being taken for the values of the incorrect samples and the number of samples for which the recursion formula is drawn up being substantially equal to the difference between the number of samples in the interval and the number of samples whose values are weighted in the weighted sum,

— determining the error energy corresponding to the error terms and averaged over the samples in the interval,

— minimizing the error energy as a function of each of the weighting factors.

The weighting factors are thus determined by determining the error term of the recursion formula for each sample for which a recursion formula can be drawn up. In these error terms the weighting factors are the only unknowns. The recursion formula is best fitting if the error energy averaged over the interval is minimal. The weighting factors are now found by minimizing the expression for the average error energy as a function of the weighting factors.

Minimizing the error energy requires the solution of a system of equations of the form:

see formula (20).

For the computation it is advantageous to replace said system by a Toeplitz system of the form:

see formula (21).

In order to obtain the Toeplitz-system an estimator must be taken for the auto correlation coefficients. A selection from different estimators is possible. A suitable estimator which yields accurate interpolation results is given by:

see formula (7).

For the computation of these estimators the values of every two samples must be multiplied by one another. In the Compact Disc Digital Audio System the values of the samples are defined by a 16-bit number, so that two 16-bit numbers must be multiplied, which may require a considerable computation time. According to the invention this computation time may be reduced if for calculating the estimators of the autocorrelation coefficients the values of the samples are described with a lower degree of quantization than the actual degree of quantization of the samples. It has been found that this coarser degree of quantization has a small and generally negligible influence in the final results of the interpolation method. For the Compact Disc Digital Audio System it is found that samples represented by a 6-bit number still yield satisfactory results. In view of the required computation time it is also advantageous if instead of carrying out a multiplication the result of the multiplication is looked up in a table containing all the possible multiplication results. If the values of the samples are expressed as a 16-bit number a multiplication will yield a 32-bit number. The number of possible results of the multiplications is then 2^{32}, so that a very large memory will be required for storing the table. Therefore, it is advantageous to describe the values of the samples with a lower degree of quantization, because this considerably reduces the number of possible results of the multiplication and consequently the required storage capacity.

A suitable method of solving the Toeplitz-system $R_2a=b_2$ is by means of the Levinson-Durbin algorithm. This algorithm is described in the afore-mentioned article in I.E.E.E. Solving the system $R_2s=b_2$ in fact means solving a system of p equations with p unknowns. Normally, this requires a number of operations of the order of p^3. However, solving the Levinson-Durbin algorithm only requires a number of operations of the order of p^2, which means a substantial reduction of the required computation time and storage capacity. This algorithm also has the advantage that in addition to the solution for the selected maximum number of terms in the recursion formula it yields the solutions with the corresponding average error energy for all recursion formulae with a number of terms smaller than the selected number as an intermediate result. The algorithm may be stopped if the error energy becomes smaller than a preset threshold. In this way it is possible to find the recursion formula with the smallest number of terms with which the samples in the interval can be described accurately.

In addition to a preselected value for the maximum number of terms of the recursion formula this maximum value may also be selected depending on the number of incorrect samples. For example, the maximum number p of samples in the recursion formula may be selected to increase as a linear function of the number m of incorrect samples. For example, for the Compact Disc Digital Audio signal, which has been sampled with a frequency of substantially 44.1 kHz, in accordance with a further embodiment, the maximum number p of samples in the recursion formula is given by the empirically found relationship $p=3m+2$. The number N of samples of the interval depends on the number m of incorrect samples and must be comparatively large relative to this number of incorrect samples for accurate computations of the weighting factors. For the Compact Disc Digital Audio signal, in accordance with a further embodiment, the number N of samples is given by the empirically found relationship $N\approx32m$. The maximum interval length is determined by the time over which the relevant signal may be regarded as stationary for interpolation purposes. For audio signals this time is at least approximately 0.01s, which corresponds to approximately 500 samples for a Compact Disc Digital Audio signal. For these signals the method yields accurate results up to approximately 16 consecutive incorrect samples, the value of these incorrect samples being calculated by means of recursion formulae with a maximum number of approximately 50 terms. It is to be

noted that the method is not limited to consecutive incorrect samples, so-called "bursts", but is also suitable for correcting non-consecutive incorrect samples.

After the weighting factors of the recursion formula have been computed by solving the system $R_2a=b_2$ in accordance with the above method the following steps may be carried out for estimating the values of the incorrect samples:

— within the first interval in which the incorrect samples are situated a second interval is determined whose first samples is situated at least a number of samples equal to the number of samples in the recursion formula before the first incorrect sample and whose last sample is situated at least a number of samples equal to the number of samples in the recursion formula after the last incorrect sample,

— for at least the samples from the first incorrect sample up to and including the last sample in the second interval the recursion formula is drawn up with the computed weighting factors,

— the error energy averaged over the relevant samples in the interval and corresponding to the error terms of the recursion formula is determined, and

— the estimates of the incorrect samples are determined by minizing the error energy as a function of the values of the incorrect samples.

Thus, the recursion formula establishes a relationship between the value of each sample in the second interval, reckoned from the first incorrect sample, and the values of a specific number of preceding correct or incorrect samples. The values of the incorrect samples must now be selected so as to minimize the mean square error of the recursion formulae. This is achieved for those values of the incorrect samples for which the average error energy corresponding to the error terms is minimal as a function of these values. The recursion formulae drawn up for the samples may be written as a system of equations of the form:

see formula (10).

Computation is simplified if this system is converted into a system of the form:

see formula (12).

This enables the error energy to be expressed as an equation of the form:

see formula (13),

for which, as is known, the vector x_{min} which minimizes this equation satisfies:

see formula (14).

The values of the samples to be corrected are thus estimated by calculating the vector x_{min}. It can be demonstrated that the vector x_{min} is found by solving the system:

see formula (15).

In the event of a so-called burst, for which $S_{t_1}, \ldots S_{t_m}$ are consecutive samples, the system $Vx_{min}=w$ is a Toeplitz system which in accordance with the invention can be solved in an advantageous manner by the Levinson algorithm, which is described in the article "The Wiener RMS error criterion in filter design and prediction", J. Math. Phys., Vol. 25, No. 4, pages 261—278, 1947. Calculating the factor x_{min} yields estimates for the values of the incorrect samples $S_{t_1}, \ldots;. S_{t_m}$ to be corrected, which in principle solves the interpolation problem.

The method in accordance with the invention may be further characterized in that the method is repeated at least one time, the weighting factors of the best-fitting recursion formula being computed from the values of the correct samples in the interval and the values of the incorrect samples computed by means of the method. This results in even more accurate values for the incorrect samples. Another advantage of this iterative method is that it enables a substantially larger number of incorrect samples within the same interval and also a larger number of incorrect samples in a larger interval to be corrected than if the method is not repeated. For the Compact Disc Digital Audio System this enables a number of approximately 50 to 100 incorrect samples to be corrected within an interval of approximately 512 to 1024 samples, the recursion length being approximately 50 to 100.

A device for carrying out the method is characterized in that the device comprises:

— input means for entering the values of samples of the samples signal,

— detection means for detecting incorrect samples,

— means for deriving from the number of samples having erroneous values a first sampling interval within which the samples having erroneous values are situated,

— means for deriving from the number of samples having erroneous values the number of samples whose values are weighted in the recursion formula,

— means for determining the weighting factors of the recursion formula from the recursion formulas with unknown weighting factors for a number of samples in the first sampling interval using the values of the samples in that interval, a first estimate being taken for the values of the samples having erroneous values,

— means for determining from the number of samples whose values are weighted in the recursion formula a second sampling interval within a first sampling interval and within which the samples having erroneous values are situated,

— means for deriving estimated values for the samples having erroneous values from the recursion formulas with known weighting factors for a number of samples in the second interval using the values of the correct samples in that interval, and

— output means for feeding out the values of samples of the sampled signal.

If the signal is a Compact Disc audio signal the number of samples in the interval can be calculated for

example by means of the formula N=32 and the maximum number of samples in the weighted sum can be calculated for example by means of the formula p=3m+2. After the interval and the recursion length have been determined the weighting factors of the recursion formula may be calculated by further determining means. In a further embodiment these means may be provided with:

— means for computing the estimators for the autocorrelation coefficients:
see formula (7), and
— means for solving the system:
see formula (16).

The computing time and storage capacitor required for computing the estimators for the autocorrelation coefficients can be reduced substantially if, in a further embodiment said computing means comprise means for reducing the degree of quantization of the samples. This reduction may be achieved for example by shifting the bits representing the values of the samples by a plurality of bit positions towards the least significant bit. In a further embodiment the computing time can also be reduced in that not every multiplication is carried out but that the means for computing the autocorrelations coefficients comprise means for determining the product of the values of two samples by means of a table of possible values of this product. The means for solving the system $Ra=b$ preferably comprise the Levinson-Durbin algorithm. A version of the device is characterized in that it comprises

— means for computing the numbers:
see formula (17),
— and means for computing the numbers:
see formula (18).

In a further version the device comprises further means for solving the system:
see formula (19).

If $S_{t_1}, \ldots S_{t_m}$ are consecutive samples, the system $vx_{min}=w$ is a Toeplitz system, which in a further embodiment is preferably solved by means comprising the Levinson algorithm.

The invention will now be described in more detail, by way of example, with reference to the accompanying drawings, in which:

Figure 1 represents an interval of a sampled signal to explain the method in accordance with the invention,
Figure 2 shows a sub-interval of the interval in Figure 1,
Figure 3 shows an apparatus for carrying out the method in accordance with the invention,
Figure 4 is a flow chart of the input/output program, and
Figure 5 is a flow chart of the interpolation program.

The method in accordance with the invention will be explained with reference to Figure 1, which shows an interval of an analogue signal which has been sampled at the instants 0, 1, . . ., N−1 with associated sampling values $S_0, S_1, \ldots S_{N-1}$. In this interval the samples detected by the detection means at the instants $0<t_1<t_2 \ldots <t_m<N-1$ have incorrect values. For these samples estimated values which as accurately as possible approximate the actual values have to be found. The method employed for this purpose is based on the recognition of the fact that for signals with comparatively slow spectral variations, such as audio and speech signals, the value of a sample in the interval may be defined by means of a best-fitting finite recursion, *i.e.* as a weighted sum of the values of a number of preceding samples, the mean square of the difference between the value of the sample and the weighted sum being small on the average. In this respect small is to be understood to mean that the energy corresponding to these differences and averaged over the interval is small in comparison with the signal energy averaged over the interval. For the interval 0, . . ., N−1 a recursion formula may be drawn up for the samples for which $p \leq j \leq N-1$, where $p \geq 1$, which formula has the form:
see formula (1).

For estimating the recursion coefficients $a_1, \ldots, a_p$ the maximum order p of the recursion formula must be known. This maximum order p may be selected depending on the number of incorrect samples. For an audio signal which originates from a Compact Disc and which comprises 44.1×10^3 samples per second, it has been found empirically that for interpolation purposes the values of the samples can be described satisfactorily by means of a recursion formula whose maximum order p is given by p=3m+2, where m is the number of incorrect samples. Later in the description it will be explained why the "maximum" order p is discussed. The recursion coefficients are determined by taking a first estimate for the values of the incorrect samples in the interval, for example by assuming that these values are zero and by drawing up a recursion formula of the order p in accordance with formula 1 for the samples for which this is possible. Subsequently, the error energy averaged over the interval and corresponding to the error terms e_j is computed. This averaged error energy Q is given by:
see formula (2),
where e_j is given by formula (1). In order to allow the assumption that the incorrect samples are zero, the number of correct samples in the interval must be comparatively large with respect to the number of incorrect samples. For the Compact Disc digital audio signal it has been found empirically that an interval with a number N of samples given by N=32m is adequate. The recursion formula is best-fitting if the error

energy Q is minimal. The recursion coefficients must then comply with $\partial Q/\partial a_i=0$ where i=1, . . ., p. Using formulae 1 and 2, this results in a system of equations given by:

see formula (3).

With the number $r_{i,k}$, which are defined as

see formula (4),

the system of equation (3) may be reduced to a system of the matrix form:

see formula (5).

A further examination of equation 4 shows that the elements $r_{i+1,k+1}$ may be written as:

see formula (6).

Since the number of samples N in the interval is large the second term is negligible, so that $r_{i+1,k+1}\approx r_{i,k}$. Moreover, the $r_{i,k}$'s of equation 4 are found to be estimators for autocovariance coefficients, which may be replaced by simpler to compute estimators for the autocorrelation coefficients. An estimator for the autocorrelation coefficients yielding satisfactory interpolation results is given by:

see formula (7).

It is to be noted that alternatively other estimators may be used for the autocorrelation coefficients. Moreover, the estimators for the autocorrelation coefficients may be computed in a manner other than that described, for example by means of Fourier transforms. By means of equations 6 and 7 the system of equation 5 may be reduced to a so-called Toeplitz system, which is given by:

see formula (8).

Here, the matrix R is a symmetric Toeplitz matrix whose elements on each diagonal are identical. The system of p equations with p unknowns, as given by equation 8, can be solved in an advantageous manner by means of the so-called Levinson-Durbin algorithm described in the aforementioned article in Proc. of the IEEE, Vol. 63, No. 4, April 1975, 561—580. This algorithm has the advantage that the system of equation 8 is solved by means of approximately p^2 operations, whilst generally this demands $\frac{1}{3}p^3$ operations. In addition to the solution for the maximum value of the order p, the algorithm also yields all solutions for 1 to p with the corresponding average error energy as intermediate results. The maximum order p which follows from the empirically found formula p=3m+2 is generally needed in order to obtain sufficiently accurate interpolation results, *i.e.* in order to obtain a sufficiently low average error energy. For very simple audio signals, such as a simple sinewave signal, this sufficiently low average error energy may already be obtained for an order smaller than the maximum order p that follows from the empirical formula. The algorithm then stops if this predetermined threshold for the average error energy is reached. A further advantage of this algorithm is that this algorithm may be employed in order to find the order p of the recursion if the maximum value of the order p is not given by an empirically found formula. If a specific value is selected for the order p this algorithm, as already stated, yields the solutions for 1 to p inclusive in increasing order with the corresonding average error energy Q as intermediate results. The algorithm may now be stopped if the error energy decreases below a predetermined threshold, the corresponding p then being the recursion order.

Solving the system of equation 8 yields the coefficients $a_1, \ldots a_p$ ($a_0=1$) for which $a_0S_j+a_1S_{j-1}+\ldots+a_pS_{j-p}$ is minimal in comparison with the average signal energy. The values of the incorrect samples are now computed by means of this equation. For this purpose a second interval $0, \ldots N'-1$ within the interval $0, \ldots, N-1$ is considered, the first sample of this second interval being situated at least p samples before the first incorrect sample t_1 and the last sample being situated at least p samples after the last incorrect sample t_m (see Figure 2). For each sample in the interval $0, \ldots, N'-1$ for which $p\leq j\leq N'-1$ a recursion formula with the recursion coefficients a_0, a_1, a_p which are now known is drawn up:

see formula (9).

The recursion equations for all samples in the interval may be written as follows in matrix form:

see formula (10).

The values for the incorrect samples $S_{t_1}, S_{t_2}, \ldots S_{t_m}$ are noted as a vector $x=(x_1, \ldots x_m)^T$. For the vector x that vector is selected for which the error energy Q averaged over the interval and given by:

see formula (11),

is minimal. The corresponding values $x_1, \ldots, x_m$ are then the estimates for the incorrect samples. By appropriate partitioning of M and *s* the system of equation (10) may be written as:

see formula (12).

In the above equation *x* is the m-vector of unknown sample values, *y* the (N'−m) vector of known sample values, A the (N'−p)×m matrix of which the j^{th} column is identical to the t_j^{th} column of M, and B the (N'−p)×(N'−m) matrix, which comprises the other columns of M. By means of equation (12) equation (11), which gives the average error energy, may be reduced to:

see formula (13).

It can be demonstrated that the vector *x* for which e^Te is minimal, complies with:

see formula (14).

Therefore, the computation of the missing samples is in a fact a computation of the vector x_{min}. By elaborating the matrices A^TA and A^TB and by rearranging equation 14, the computation of x_{min} in effect requires the solution of the system:

see formula (15).

In the present example the incorrect samples form a so-called burst, *i.e.* the incorrect samples form a sequence of consecutive samples. In the present case the symmetric matrix A^TA is a Toeplitz matrix, so that the system of equation 15 is a Toeplitz system, which can be solved advantageously by means of the Levinson algorithm. It is to be noted that the method is not limited to so-called bursts, but that the values of a plurality of non-consecutive incorrect samples can also be corrected by means of the method. The system of equation Z(15) may then be solved for example by means of an LU-decomposition.

When the above method is applied to signals of the Compact Disc Digital Audio System it is found possible to compute burst errors in an accurate manner up to approximately 16 incorrect samples. In the case of 16 incorrect samples an interval of approximately 512 samples is required for determining the recursion coefficients, which corresponds to a period of approximately 0.01 s. Within this period audio signals may be regarded as stationary for interpolation purposes.

In a second version of a method in accordance with the invention the values of the samples are computed in an iterative fashion. Firstly, the samples are computed in the above manner, assuming that the values of the incorrect samples are zero for computing the estimators for the autocorrelation coefficients in accordance with equation 7. Subsequently, the values of the incorrect samples are recalculated, the calculated values now being assigned to the incorrect samples in order to compute the autocorrelation coefficients. This step may be repeated several times. This may lead to a higher accuracy of the computed values of the incorrect samples. By means of this iterative method it is alternatively possible to select a smaller interval in order to compute an equal number of incorrect samples. Thus, for a burst error of 16 samples an interval of 100 instead of 512 samples is adequate, the same accuracy of the computed values of the incorrect samples being attained after, for example, 2 or 3 iteration steps. Moreover, this iterative method enables a larger number of incorrect samples to be corrected within comparable intervals than in the case that the method is not repeated. In an interval of approximately 1024 samples this enables approximately 100 incorrect samples to be corrected.

Figure 3 is the block diagram of a device in accordance with the invention. The block 1 is a compact-disc player as described comprehensively in Philips Techn. Review 40, No. 9, 1981/1982. In this respect it is to be noted that the digital audio signal to which error-correction codes have been added and which has been time-interleaved in accordance with a specific code, is recorded on a disc along a spiral track of pits and intermediate areas. The digital information is read by means of a laser and is subsequently applied to a signal processing unit 2, in which this digital signal is demodulated and, if necessary, corrected. On output 3 the values of the samples appear in the form of 16-bit numbers. If more errors occur than can be corrected by means of the error-correction codes, these errors are merely detected. If a sample with an incorrect value appears on output 3 an error-flag will appear on output 4 of the processing unit, which flag is formed by a 1-bit signal. The values of the samples and the error-flags are applied to the respective input devices 6 and 7 of a microprocessor arrangement 5 by means of which the incorrect values are corrected. In addition to an output device 8 for feeding out the sample values this processor 5 comprises a central processing unit 9 with a control unit and an arithmetic and logic unit. Moreover, the processor 5 comprises three memories 10, 11 and 12. The memories 10 and 11 are RAMs, the memory 10 serving as a cyclic buffer and the memory 11 as a working store for the storage of intermediate results. The memory 12 is a ROM in which the program for correcting the incorrect values of the samples are stored. The memories 10, 11 and 12 are coupled to the central processing unit 9 by a data bus 13, *via* which bus the data can be transferred. For the transfer of addresses the memories 10, 11 and 12 and the input and output devices 6, 7 and 8 are coupled to the central processing unit 9 *via* an address bus 14. Moreover, the processor 5 is provided with a clock 15 by means of which the operation of the central processing unit is interrupted to allow the input and output of data. For digital audio a constant transfer of samples is required in order to ensure that the samples appear on the output with the same frequency as that with which the signal has been sampled. The input and output of data is controlled by a program stored in the program store 12. Figure 4 is a flow chart of this program. The program may be described as follows: for the inscriptions, see Table 1.

- block 41, inscription: "wait for interrupt", description: data input and output is possible during an interrupt, *i.e.* when a clock pulse from the data clock 15 appears.
- block 42, inscription: "output/input", description: when a clock pulse appears the value of a new sample is entered *via* input 7 and is written into the first free address in the memory 10, the oldest sample in the memory 10 being read out and fed out.
- block 43, inscription: "input error?", description: when a new sample is read in the error flag is read in *via* input 6. If no incorrect value is detected the next clock pulse is awaited to repeat the procedure.
- block 44, inscription: "update error table", description: if a sample with an incorrect value is detected the address of this sample in the memory 10 is written into a table of incorrect samples in the working store 11. In the above manner the cyclic buffer memory 10 functions as a delay line whose delay depends on the time required for one interpolation, measured from the entry of the first incorrect sample up to and including the correction of the last incorrect sample. The program stored in the memory 12, by means of which programme the values of the incorrect samples are calculated by interpolation, is started

if the input/output program detects an incorrect sample. This interpolation program is explained with reference to Figure 5, which is a flow chart of the program (for inscription: see Table 2).

— block 51, inscription: "wait while error table empty", description: as long as no error is detected the interpolation program is inoperative.

— block 52, inscription: "compute m, N(m), p(m)", description: when an incorrect sample is detected the number N of samples in the interval and the maximum number p of samples in the weighted sum of the recursion formula are computed with respective formulae N=32 and p=13m+2, where m is the number of incorrect samples. For m=1 this yields N=32 and p=5. This means, that prior to and subsequently to the incorrect sample at position t_1 each time 16 correct samples must be present. The 16 preceding samples are correct because it concerns the first incorrect sample that has been detected. If the 16 next samples are correct, the program is continued. If within these 16 samples at position t_2 a second incorrect sample is detected, N=64 and p=8 if m=2, so that 32 correct samples must appear before and after the sample t_2. This may continue until a specific maximum interval length is reached. This maximum is attained for example in the case of 16 consecutive errors for N=512, which corresponds to the period over which audio signals may be regarded as stationary.

— block 53, inscription: "wait until (Sj) j=0, . . ., N−1 are in buffer". Description: If m, N and p are known, the values of the N samples of the interval are stored in a part of the memory 11, the values of the incorrect samples being assumed to be equal to zero.

— block 54, inscription: "compute r(j), j=0, . . ., p". With the values of the samples in this interval the autocorrelation coefficients r(j) are calculated, which are given by:

see formula (7).

The computation of the products S_kS_{k+j} can be effected in a rapid manner by reducing the degree of quantization of the samples. This may be achieved for example by shifting the 16-bit numbers representing the sample values by a plurality of bits towards the least significant bit. It has been found that this lower degree of quantization has a minor influence on the ultimately computed values of the incorrect samples. Thus, satisfactory results can be achieved for samples whose values are expressed by a 6-bit number. The products S_k+S_{k+j} can also be obtained rapidly by not repeating each calculation but by looking up the results, in a table in the working store 11, which table contains all possible results of the product S_kS_{k+j}.

— block 55, inscription: "compute $a_1, \ldots a_p$". Description: by means of the calculated autocorrelation coefficients the recursion coefficients $a_1 \ldots a_p$ are computed by solving the following system by means of the Levinson-Durbin algorithm:

see formula (16).

— block 56, inscription: "compute $\lambda_0, \ldots, \lambda_p$", description: by means of the recursion coefficients now known the factors λ_k are calculated, these factors being given by:

see formula (17).

— block 57, inscription "compute syndrome". Subsequently, the components w_l of the vector w are calculated, which components are given by:

see formula (18).

Summation is applied over an interval 0, . . ., N'−1, of which the first sample S_0 is situated p samples before the first incorrect sample S_{t_1} and the last sample $S_{N'-1}$ is situated p samples after the last incorrect sample S_{t_m}.

— block 58, inscription: "solve missing samples". Description: the values of the incorrect samples $S_{t_1}, \ldots S_{t_m}$ are computed by solving the system:

see formula (19).

The system $Vx_{min}=w$ is solved with a program for solving a system of m equations with m unknowns, If $S_{t_1}, \ldots S_{t_m}$ are consecutive samples, this system is a Toeplitz system, which can be solved by means of the Levinson algorithm.

— block 59, the values of the incorrect samples which were set to zero in the memory 10 are now replaced by the estimates $S_{t_1}, \ldots S_{t_m}$. After this program is repeated.

The invention is not limited to the example described in the foregoing. For example, the calculated values $S_{t_1}, \ldots S_{t_m}$ may be employed for recomputing the autocorrelation coefficients r(j), the recomputed values now being adopted for the values of the incorrect samples. Subsequently, the part 60 of the program indicated by a broken line in Figure 5 is repeated. It may be repeated several times. In this way a higher accuracy of the calculated values can be achieved. Moreover, if this iterative method is applied, the number N of samples may be reduced in order to compute the same number of incorrect samples with the same accuracy. Also, this iterative method enables a larger number of incorrect samples to be corrected within the same interval and even within smaller intervals. It will be evident that within the scope of the present invention several devices are possible by means of which the values of incorrect samples can be corrected with the specified interpolation method based on best-fitting finite recursions. The invention is not limited to the correction of incorrect samples of digital audio or speech signals, but may also be used for example

for suppressing scratches on grammophone records, in which case the analog signal obtained from the record must be sampled first.

Formulae:

(1) $e_j = a_0 S_j + a_1 S_{j-1} + \ldots + a_p S_{j-p}$

where:
— $a_0, a_1, \ldots a_p$, in which $a_0 = 1$ are the recursion coefficients.
— $S_j \ldots, S_{j-p}$ are the values of the samples at the instants $j, \ldots, j-p$, and
— e_j is the error term.

(2) $$Q = \frac{1}{N-p} \sum_{j=p}^{N-1} |e_j|^2$$

(3) $$\sum_{k=0}^{p} a_k \left(\frac{1}{N-p} \sum_{j=p}^{N-1} S_{j-k} S_{j-l} \right) = 0$$

(4) $$r_{l,k} = \frac{1}{N-p} \sum_{j=p}^{N-1} S_{j-k} S_{j-l};\ i, k = 0, 1, \ldots p$$

(5) $Ra = b$

— where $R = (r_{l,k})\ i, k = 1, \ldots, p$
$a = [a_1, \ldots, a_p]^T$
$b = [-r_{l,0}, \ldots, -r_{p,0}]^T$

(6) $$r_{l+1,k+1} = r_{l,k} + \frac{1}{N-p}(S_{p-k-1} S_{p-l-1} - S_{N-k-1} S_{N-l-1}).$$

(7) $$r(j) = \frac{1}{N} \sum_{k=0}^{N-j-1} S_k S_{k+j}$$

where $j = (i-k) = 0, \ldots p$
$i, k = 0, \ldots p$

(8) $$\begin{bmatrix} r(0) & r(1) & \ldots & r(p-1) \\ r(1) & & & \\ \vdots & & & \\ r(p-1) & r(p-2) & & r(0) \end{bmatrix} \begin{bmatrix} a_1 \\ \vdots \\ a_p \end{bmatrix} \equiv \begin{bmatrix} r(1) \\ \vdots \\ r(p) \end{bmatrix}$$

(9) $e_j = a_0 S_j + a_1 S_{j-1} + \ldots + a_p S_{j-p}$

(10) $MS = e$

— where $M = (m_{ij})\ i = 0, \ldots N'-p-1$
$= (a_{p-i+j})\ i = 0, \ldots N'-p-1$
$j = 0, \ldots N'-1$
$j = 0, \ldots N'-1$

where N'=the number of samples in the second interval, $a_i=0$, for $i<0$ and $i>p$,
and $a_0=1$, $a_1, \ldots, a_p$=the weighting factors of the recursion formula
— where $S=(S_0, \ldots, S_{N'-1})^T$ with $S_0, \ldots, S_{N'-1}$ being the values of the samples in the second interval, and
— where $e(e_p, e_{p+1}, \ldots, e_{N'-1})^T$ with $e_p, \ldots, e_{N'-1}$ being the values of the error terms of the recursion formulas for the samples $S_p, \ldots S_{N'-1}$.

$$(11) \quad Q=\frac{1}{N-p}\sum_{j=p}^{N'-1}|e_j|^2$$

$$(12) \quad Ax+By=e$$

— where $A=(a_p+t_j-i)$ $i=0, \ldots, N'-p-1$
$j=1, \ldots, m$
$x=(S_{t_1}, \ldots S_{t_m})^T$ where $S_{t_1}, \ldots S_{t_m}$ are the values of the incorrect samples at positions $t_1, \ldots t_m$ in the second interval
$B=(a_{p+j-i})$ $i=0, \ldots, N'-p-1$
$j=1, \ldots, N'-1$; $j=t_1, \ldots, t_m$

— and where $y=[S_k]^T$ $k=0, \ldots N'-1$; $k\neq t_1, \ldots, t_m$

$$(13) \quad e^Te=(y^TB^T+x^TA^T)(Ax+By)$$

$$(14) \quad (A^TA)x_{min}=-A^TBy$$

$$(15) \quad Vx_{min}=w$$

— where $V=A^TA$ with $(A^TA)_{i,j}=\lambda_{t_i-t_j}$ in which $i, j=1, \ldots, m$

— where $w=(A^TB)y$ with $(A^TBy)_i=\sum_{k=0}^{N'-1}\lambda_{t_i-k}S_k$

in which $i=1, \ldots, m$
$k\neq t_1, \ldots, t_m$

with $(A^TB)_{i,k}=\lambda_{t_i-k}$

in which $\begin{cases} i=1, \ldots, m \\ k\neq t_1, \ldots t_m \end{cases}$

— and where $\lambda_k=\sum_{l=0}^{p}a_la_{l+k}$

with $\begin{cases} k=-p, \ldots, 0, \ldots p \\ a_l=0, \text{ for } l>p \text{ and } l<0 \end{cases}$

$$(16) \quad Ra=b$$

— where $r_{i,k}=r(j)$ with $j=(i-k)=0, \ldots, p-1$; $i, k=1, \ldots, p$
$b=-[r(1), r(2), \ldots, r(p)]^T$
$a=[a_1, \ldots, a_p]^T$

where a_l are the weighting factors

$$(17) \quad \lambda_k=\sum_{l=0}^{p}a_la_{l+k}$$

where $k=-p, \ldots, 0, \ldots p$ and $a_l=0$ for $l>p$ and $l<0$

$$(18)\quad w_i=(A^TBy)_i=\sum_{k=0}^{N'-1}\lambda_{t_i-k}S_k$$

where $i=1,\ldots,m$
$k\neq t_1,\ldots t_m$

$$(19)\quad Vx_{min}=w$$

— where $V=A^TA$ with $(A^TA)_{i,j}=\lambda_{t_i-t_j}$

in which $i, j=1,\ldots,m$
$x_{min}=[S_{t_1},\ldots,S_{t_m}]^T$

— where $S_{t_1},\ldots S_{t_m}$ are the values of the missing samples

$$(20)\quad R_1a=b_1$$

— where $R_1=r_{i,k}=\frac{1}{N-p}\sum_{j=p}^{N-1}S_{j-k}S_{j-i}$

in which $i, k=1,\ldots,p$
N=the number of samples in the interval
p=the number of samples in the recursion formula
S_{j-k}=the value of the $(j-k)^{th}$ sample in the interval
$r_{i,k}$=the (i, k) the estimated autocovariance coefficient

— where $a_1=[a_1,\ldots,a_p]^T$ in which $a_1,\ldots a_p$ are the weighting factors of the recursion formula, and
— where $b=[-r_{1,0},\ldots,-r_{p,0}]^T$

in which $r_{i,0}=\frac{1}{N-p}\sum_{j=p}^{N-1}S_jS_{j-i}$ with $i=1,\ldots p$

$$(21)\quad R_2a=b_2$$

— in which R_1 has been replaced by $R_2=(r(i-k))_{i,k}$ where $i, k=1,\ldots p$
— in which $r(i-k)$ is an estimator for the $(i-k)^{th}$ autocorrelation coefficient, and
— where the vector b_1 has been replaced by the vector $b_2=-[r(1), r(2),\ldots, r(p)]$

where $r(i)$ is an i^{th} estimator for the autocorrelation coefficient.

TABLES:

TABLE 1:
Inscriptions flow chart Figure 4

Block number	Inscription
41	"wait for interrupt"
42	"output/input"
43	"input error?"
44	"update error table"

TABLE 2
Inscriptions flow chart Figure 5

Block number	Inscription
51	"wait while error table empty"
52	"compute m N(m), p(m)"
53	"wait until (S_j), j=0, . . . N−1 are in buffer"
54	"compute r(j), j=0, . . . p"
55	"compute $a_1, \ldots, a_p$"
56	"compute $\lambda_0, \ldots \lambda_p$"
57	"compute syndrome"
58	"solve missing samples"
59	"update buffer"

Claims

1. A method of correcting erroneous values of samples of an equidistantly samples signal by means of a recursion formula expressing the erroneous samples as a weighted sum of the values of a number of preceding samples, characterized by the steps of
— deriving from the number of samples having erroneous values a first sampling interval within which the samples having erroneous values are situated,
— deriving from the number of samples having erroneous values the number of samples whose values are weighted in the recursion formula,
— detecting the weighting factors of the recursion formula from the recursion formulas with unknown weighting factors for a number of samples in the first sampling interval using the values of the samples in that interval, a first estimate being taken for the values of the samples having erroneous values,
— determining from the number of samples whose values are weighted in the recursion formula a second sampling interval within the first sampling interval and within which the samples having erroneous values are situated,
— deriving estimated values for the samples having erroneous values from the recursion formulas with known weighting factors for a number of samples in the second interval using the values of the correct samples in that interval.

2. A method as claimed in Claim 1, characterized in that for determining the weighting factors of the weighted sum of the recursion formula the following steps are carried out:
— drawing up a recursions formula for a number of samples within the interval, which formula contains a weighted sum of the values of a number of preceding samples, the difference between the value of a sample and the value of the weighted sum being expressed as an error term, a first estimate being taken for the values of the incorrect samples and the number of samples for which a recursion formula is drawn up being substantially equal to the difference between the number of samples in the interval and the number of samples whose values are weighted in the weighted sum,
— determining the error energy corresponding to the error terms and averaged over the samples in the interval, and
— minimizing the error energy as a function of each of the weighting factors.

3. A method as claimed in Claim 1 or 2, characterized in that for determining the weighting factors of the weighted sum of the recursion formula the system of equations to be solved of the form: see formula (20), is replaced by a Toeplitz system of the form:
see formula (21).

4. A method as claimed in Claim 3, characterized in that the estimators for the autocorrelation coefficients are given by:
see formula (7).

5. A method as claimed in Claim 3 or 4, characterized in that for computing the estimators for the autocorrelation coefficients the values of the samples are described with a lower degree of quantization than the actual degree of quantization of the samples.

6. A method as claimed in Claim 4 or 5, characterized in that for calculating the estimators of the autocorrelations coefficients the result of a multiplication of the values of two samples is looked up in a table containing the possible results of the multiplication.

7. A method as claimed in Claim 3, 4, 5 or 6, characterized in that the Toeplitz system $R_2 a = b_2$ is solved by means of the Levinson-Durbin algorithm.

8. A method as claimed in Claim 1, characterized in that a predetermined value is adopted for the maximum number p of samples in the recursion formula.

9. A method as claimed in Claim 1, characterized in that maximum number p of samples of the recursion depends on the number m of incorrect samples.

10. A method as claimed in Claim 9, characterized in that the maximum number p of samples increases as a linear function of the number of incorrect samples.

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TABLE 2
Inscriptions flow chart Figure 5

Block number	Inscription
51	"wait while error table empty"
52	"compute m N(m), p(m)"
53	"wait until (S_j), j=0, ... N−1 are in buffer"
54	"compute r(j), j=0, ... p"
55	"compute $a_1, \ldots, a_p$"
56	"compute $\lambda_0, \ldots \lambda_p$"
57	"compute syndrome"
58	"solve missing samples"
59	"update buffer"

Claims

1. A method of correcting erroneous values of samples of an equidistantly samples signal by means of a recursion formula expressing the erroneous samples as a weighted sum of the values of a number of preceding samples, characterized by the steps of

— deriving from the number of samples having erroneous values a first sampling interval within which the samples having erroneous values are situated,

— deriving from the number of samples having erroneous values the number of samples whose values are weighted in the recursion formula,

— detecting the weighting factors of the recursion formula from the recursion formulas with unknown weighting factors for a number of samples in the first sampling interval using the values of the samples in that interval, a first estimate being taken for the values of the samples having erroneous values,

— determining from the number of samples whose values are weighted in the recursion formula a second sampling interval within the first sampling interval and within which the samples having erroneous values are situated,

— deriving estimated values for the samples having erroneous values from the recursion formulas with known weighting factors for a number of samples in the second interval using the values of the correct samples in that interval.

2. A method as claimed in Claim 1, characterized in that for determining the weighting factors of the weighted sum of the recursion formula the following steps are carried out:

— drawing up a recursions formula for a number of samples within the interval, which formula contains a weighted sum of the values of a number of preceding samples, the difference between the value of a sample and the value of the weighted sum being expressed as an error term, a first estimate being taken for the values of the incorrect samples and the number of samples for which a recursion formula is drawn up being substantially equal to the difference between the number of samples in the interval and the number of samples whose values are weighted in the weighted sum,

— determining the error energy corresponding to the error terms and averaged over the samples in the interval, and

— minimizing the error energy as a function of each of the weighting factors.

3. A method as claimed in Claim 1 or 2, characterized in that for determining the weighting factors of the weighted sum of the recursion formula the system of equations to be solved of the form: see formula (20), is replaced by a Toeplitz system of the form:

see formula (21).

4. A method as claimed in Claim 3, characterized in that the estimators for the autocorrelation coefficients are given by:

see formula (7).

5. A method as claimed in Claim 3 or 4, characterized in that for computing the estimators for the autocorrelation coefficients the values of the samples are described with a lower degree of quantization than the actual degree of quantization of the samples.

6. A method as claimed in Claim 4 or 5, characterized in that for calculating the estimators of the autocorrelations coefficients the result of a multiplication of the values of two samples is looked up in a table containing the possible results of the multiplication.

7. A method as claimed in Claim 3, 4, 5 or 6, characterized in that the Toeplitz system $R_2 a = b_2$ is solved by means of the Levinson-Durbin algorithm.

8. A method as claimed in Claim 1, characterized in that a predetermined value is adopted for the maximum number p of samples in the recursion formula.

9. A method as claimed in Claim 1, characterized in that maximum number p of samples of the recursion depends on the number m of incorrect samples.

10. A method as claimed in Claim 9, characterized in that the maximum number p of samples increases as a linear function of the number of incorrect samples.

11. A method as claimed in Claim 10, characterized in that the signal is an audio signal which is sampled with a frequency of substantially 44.1 kHz and the relationship between the maximum number p of samples of the recursion and the number m of incorrect samples is given by $p \approx 3m+2$.

12. A method as claimed in Claim 1, characterized in that the number N of samples in the interval is determined by the number m of incorrect samples.

13. A method as claimed in Claim 12, characterized in that the number N of samples increases as a linear function of the number m of incorrect samples.

14. A method as claimed in Claim 10, characterized in that the signal is an audio signal which is sampled with a frequency of substantially 44.1 kHz and the relationship between the number N of samples in the interval and the number of incorrect samples is given $N \approx 32m$.

15. A method as claimed in any one of the preceding Claims, characterized in that for estimating the values of the incorrect samples the following steps are carried out:

— within the first interval in which the incorrect samples are situated a second interval is determined whose first sample is situated at least a number of samples equal to the number of samples in the recursion formula before the first incorrect sample and whose last sample is situated at least a number of samples equal to the number of samples in the recursion formula after the last incorrect sample,

— for at least the samples from the first incorrect sample up to and including the last sample in the second interval the recursion formula is drawn up with the computed weighting factors,

— the error energy which averaged over the relevant samples in the interval and corresponding to the error terms of the recursion formulas is determined, and

— the estimates of the incorrect samples are determined by minizing the error energy as a function of the values of the incorrect samples.

16. A method as claimed in Claim 15, characterized in that the recursion formulae constitute the system of recursion equations of the form: see formula (10), and this system is converted into a system of the form: see formula (12).

17. A method as claimed in Claim 16, characterized in that by means of the system $Ax+By=e$ the equation for the error energy averaged over the relevant samples in the interval is converted into an equation of the form:

see formula (13),

in which the vector x_{min} for which the error energy is minimal is given by:

see formula (14).

18. A method as claimed in Claim 17, characterized in that the vector x_{min} is computed by solving the system:

see formula (15).

19. A method as claimed in Claim 18, characterized in that the incorrect samples are situated at consecutive instants $t_1, \ldots t_m$ and the system $Vx_{min}=w$ is solved by means of the Levinson algorithm.

20. A method as claimed in any one of the preceding Claims, characterized in that the method is repeated at least one time, the weighting factors of the best-fitting recursion formula being computed from the values of the correct samples in the interval and the values of the incorrect samples as estimated by means of the method.

21. A device for carrying out the method as claimed in Claim 1, characterized in that the device comprises:

— input means for entering the values of samples of the sampled signal,

— detection means for detecting incorrect samples,

— means for deriving from the number of samples having erroneous values a first sampling interval within which the samples having erroneous values are situated,

— means for deriving from the number of samples having erroneous values the number of samples whose values are weighted in the recursion formula,

— means for determining the weighting factors of the recursion formula from the recursion formulas with unknown weighting factors for a number of samples in the first sampling interval using the values of the samples in that interval, a first estimate being taken for the values of the samples having erroneous values,

— means for determining from the number of samples whose values are weighted in the recursion formula a second sampling interval within the first sampling interval and within which the samples having erroneous values are situated,

— means for deriving estimated values for the samples having erroneous values from the recursion formulas with known weighting factors for a number of samples in the second interval using the values of the correct samples in that interval, and

— output means for feeding out the values of samples of the sampled signal.

22. A device as claimed in Claim 21, characterized in that the means for determining the weighting factors are provided with:

— means for computing the autocorrelation coefficients:
see formula (7), and

— means for solving the system:
see formula (16).

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23. A device as claimed in Claim 22, characterized in that the means for computing the autocorrelation coefficients comprise means for reducing the degree of quantization of the values of the samples.

24. A device as claimed in Claim 22, characterized in that the means for computing the autocorrelation coefficients comprise means for determining the product of the values of two samples by means of a table of possible values of this product.

25. A device as claimed in Claim 22, 23 or 24, characterized in that the means for solving the system $Ra=b$ comprise the Levinson-Durbin algorithm.

26. A device as claimed in any one of the Claims 21 to 25, characterized in that it comprises:
— means for computing the numbers:
see formula (17), and
— means for computing the numbers:
see formula (18).

27. A device as claimed in Claim 26, characterized in that it further comprises means for solving the system:
see formula (19).

28. A device as claimed in Claim 27, where $S_{t_1}, \ldots S_{t_m}$ are consecutive samples, characterized in that the means for solving the system $Vx_{min}=w$ comprise a Levinson algorithm.

29. An apparatus for reproducing digital information from a carrier, in particular an optically readable record carrier on which digital information is stored in the form of optically detectable areas alternating with intermediate areas, characterized in that the apparatus is provided with a device as claimed in any one of the preceding Claims 21 to 28.

Patentansprüche

1. Verfahren zur Korrektur fehlerhafter Werte von Abtastproben eines äquidistant abgetasteten Signals mittels einer Rekursionsformel, die die fehlerhaften Abtastproben als gewogene Summe einer Anzahl vorangehender Abtastproben darstellt, dadurch gekennzeichnet, daß das Verfahren folgende Schritte umfaßt:
— Ableiten eines ersten Abtastintervalls, in dem sich die Abtastproben mit fehlerhaften Werten befinden, aus der Anzahl von Abtastproben mit fehlerhaften Werten,
— Ableiten der Anzahl von Abtastproben, deren Werte in der Rekursionsformel gewogenen werden, aus der Anzahl von Abtastproben mit fehlerhaften Werten,
— Bestimmen der Bewertungsfaktoren der Rekursionsformel aus den Rekursionsformeln mit unbekannten Bewertungsfaktoren für eine Anzahl von Abtastproben im ersten Abtastintervall unter Verwendung der Werte der Abtastproben in diesem Intervall, wobei eine erste Abschätzung für die Werte der Abfastproben mit fehlerhaften Werten gemacht wird,
— Bestimmen eines zweiten Abtastintervalls innerhalb des ersten Abtastintervalls, und in dem sich die Abtastproben mit fehlerhaften Werten befinden, aus der Anzahl von Abtastproben, deren Werte in der Rekursionsformel gewogen sind,
— Ableiten abgeschätzter Werte Für die Abtastproben mit fehlerhaften Werten aus den Rekursionsformeln mit bekannten Bewertungsfaktoren für eine Azahl von Abtastproben im zweiten Abtastintervall unter Verwendung der Werte der richtigen Abtastproben in diesem Intervall.

2. Verfahren nach Anspruch 1, dadurch gekennzeichnet, daß zum Bestimmen der Bewertungsfaktoren der gewogenen Summe der Rekursionsformel folgende Schritte durchgeführt werden:
— Aufstellen einer Rekursionsformel für eine Anzahl von Abtastproben im Intervall, wobei diese Formel eine gewogene Summe der Werte einer Anzahl vorangehender Abtastproben enthält, wobei der Unterschied zwischen dem Wert einer Abtastprobe und dem Wert der gewogenen Summe als ein Fehlerglied dargestellt wird, wobei eine erste Abschätzung für die Werte der unrichtigen Abtastproben genommen wird und die Anzahl der Abtastproben, für die eine Rekursionsformel aufgestellt wird, im wesentlichen gleich dem Unterschied zwischen der Anzahl von Abtastproben im Intervall und der Anzahl von Abtastproben ist, deren Werte in der gewogenen Summe bewertet sind,
— Bestimmen der Fehlerenergie, entsprechend den Fehlergliedern und gemittelt über Abtastproben im Intervall, und
— Minimisieren der Fehlerenergie abhängig von jedem der Bewertungsfaktoren.

3. Verfahren nach Anspruch 1 oder 2, dadurch gekennzeichnet, daß zum Bestimmen der Bewertungsfaktoren der gewogenen Summe der Rekursionsformel das aufzulösende Gleichungssystem der Formel:

$$R_1 a = b_1$$

— worin

$$R_1 = r_{i,k} = \frac{1}{N-p} \sum_{j=p}^{N-1} S_{j-k} S_{j-i}$$

— worin i, k=1 . . . p

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N=die Anzahl der Abtastproben im Intervall
p=die Anzahl. der Abtastproben in der Rekursionsformel
S_{j-k}=der Wert der (j−k). Abtastprobe im Intervall
$r_{i,k}$=der (i,k), Abgeschätzte Autokorrelationskoeffizient
— worin $a_1=[a_1 \ldots a_p]^T$ ist, worin $a_1 \ldots a_p$ die Bewertungsfaktoren der Rekursionsformel sind, und
— worin $b=[-r_{1,0} \ldots -r_{p,0}]^T$ ist, worin

$$r_{i,0}=\frac{1}{N-p}\sum_{j=p}^{N-1} S_j S_{j-1}$$

— worin i=1, ... p
durch ein Toeplitz-System nachstehender Formel ersetzt wird:

$$R_2 a=b_2$$

— worin R_1 durch $R_2=(r(i-k))_{i,k}$ ersetzt wurde, worin i, k=1 ... p ist,
— worin r(i−k) eine Schätzfunktion für den (i−k). Autokorrelationskoeffizienten ist, und
— worin der Vektor b_1 durch den Vektor $b_2=-[r(1), r(2) \ldots r(p)]$ ersetzt wurde, worin r(i) eine i. Schätzfunktion für den Autokorrelationskoeffizienten ist.

4. Verfahren nach Anspruch 3, dadurch gekennzeichnet, daß die Schätzfunktionen für die Autokorrelationskoeffizienten durch nachstehende Formel gegeben werden:

$$r(j)=\frac{1}{N}\sum_{k=0}^{N-j-1} S_k S_{k+j}$$

worin j=(i−k)=0 ... p und i, k=0 ... p sind.

5. Verfahren nach Anspruch 3 oder 4, dadurch gekennzeichnet, daß zum Berechnen der Schätzfunktionen der Autokorrelations-koeffizienten die Werte der Abtastproben mit einem niedrigeren Quanti-sierungsgrad als der wirkliche Quantisierungsgrad der Abtastproben beschrieben werden.

6. Verfahren nach Anspruch 4 oder 5, dadurch gekennzeichnet, daß zum Berechnen der Schätzfunktionen der Autokorrelations-koeffizienten das Ergebnis einer Vervielfachung der Werte von zwei Abtastproben in einer Tabelle mit den möglichen Ergebnissen der Vervielfachung aufgesucht wird.

7. Verfahren nach Anspruch 3, 4, 5 oder 6, dadurch gekennzeichnet, daß das Toeplitz-System $R_2\, a=b_2$ mit Hilfe des Levinson-Durbin-Algorithmus aufgelöst wird.

8. Verfahren nach Anspruch 1, dadurch gekennzeichnet, daß ein vorgegebener Wert für die Höchstzahl von p Abtastproben in der Rekursionsformel angenommen wird.

9. Verfahren nach Anspruch 1, dadurch gekennzeichnet, daß die Höchstzahl von p Abtastproben der Rekursion von der Anzahl von m unrichtigen Abtastproben abhängig ist.

10. Verfahren nach Anspruch 9, dadurch gekennzeichnet, daß die Höchstzahl von p Abtastproben als lineare Funktion der Anzahl unrichtiger Abtastproben größer wird.

11. Verfahren nach Anspruch 10, dadurch gekennzeichnet, daß das Signal ein mit einer Frequenz von im wesentlichen 44,1 kHz abgetastetes Audiosignal ist und die Beziehung zwischen der Höchstzahl von p Abtastproben der Rekursion und der Anzahl von m unrichtiger Abtastproben durch $p \approx 3m+2$ gegeben wird.

12. Verfahren nach Anspruch 1, dadurch gekennzeichnet, daß die Anzahl von N abtastproben im Intervall durch die Anzahl von m unrichtigen Abtastproben bestimmt wird.

13. Verfahren nach Anspruch 12, dadurch gekennzeichnet, daß die Anzahl von N abtastproben als lineare Funktion der Anzahl von m unrichtigen Abtastproben ansteigt.

14. Verfahren nach Anspruch 10, dadurch gekennzeichnet, daß das Signal ein mit einer Frequenz von im wesentlichen 44,1 kHz abgetastetes Audiosignal und die Beziehung zwischen der Anzahl von N abtastproben im Intervall und der Anzahl unrichtiger Abtastproben durch N 32m gegeben ist.

15. Verfahren nach einem oder mehreren der vorangehenden Ansprüche, dadurch gekennzeichnet, daß zum Abschätzen der Werte der unrichtigen Abtastproben folgende Schritte durchgeführt werden:
— im ersten Intervall, in dem sich die unrichtigen Abtastproben befinden, wird ein zweites Intervall bestimmt, dessen erste Abtastprobe sich wenigstens eine Anzahl von Abtastproben gleich der Anzahl von Abtastproben in der Rekursionsformel vor der ersten unrichtigen Abtastprobe und dessen letzte Abtastprobe sich wenigstens eine Anzahl von Abtastproben gleich der Anzahl von Abtastproben in der Rekursionsformel nach der letzten unrichtigen Abtastprobe befinden,
— wenigstens für die Abtastproben ab der ersten unrichtigen Abtastprobe bis zur letzten Abtastprobe im zweiten Intervall wird die Rekursionsformel mit den berechneten Bewertungsfaktoren aufgestellt wird,
— die Fehlerenergie, die über die betreffenden Abtastproben im Intervall gemittelt wird und den Fehlergliedern der Rekursionsformeln entsprechen, wird bestimmt, und
— die Abschätzungen der unrichtigen Abtastproben werden durch Mihimisieren der Fehlerenergie abhängig von den Werten der unrichtigen Abtastproben bestimmt.

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16. Verfahren nach Anspruch 15, dadurch gekennzeichnet, daß die Rekursionsformeln das System der Rekursionstleichungen in folgender Form bilden:

$$MS=e$$

— worin $M=(m_{ij})i=0,\ldots N'-p-1=(a_{p-i+j})i=0,\ldots N'-p-1$
$j=0,\ldots N'-1 \qquad j=0,\ldots N'-1$
worin N'=die Anzahl der Abtastproben im zweiten Intervall und
$a_i=0$ ist, wenn $i<0$ und $i>p$,
$a_0=1$, $a_1 \ldots a_p$=die Bewertungsfaktoren der Rekursionsformel sind,
— worin $S=(S_0 \ldots S_{N'-1})^T$ ist, wobei $S_0 \ldots S_{N'-1}$ die Werte der Abtastproben in zweiten Intervall sind, und
— worin $e=(e_p, e_{p+1} \ldots e_{N'-1})^T$ ist, wobei $e_p \ldots e_{N'-1}$ die Werte der Fehlerglieder der Rekursionsformeln für die Abtastproben $S_p \ldots S_{N'-1}$ sind,
und dieses System wird in eine System folgender Form umgesetzt:

$$Ax+By=e$$

— worin $A=(a_{p+t_{j-i}})$ $i=0,\ldots, N'-p-1$
$j=1,\ldots, m$
$x=(S_{t_1},\ldots S_{t_m})^T$ worin $S_{t_1},\ldots, S_{t_m}$
die Werte der unrichtigen Abtastproben an den Stellen $t_1 \ldots t_m$ im zweiten Intervall sind,
— und worin $B=(a_{p+j-i})$ $i=0,\ldots N'-p-1$
$j=1,\ldots, N'-1;\ j\neq t_1,\ldots, t_m$
— und worin $y=[S_k]^T$ $k=0\ldots N'-1$ und $k\neq t_1 \ldots t_m$ sind.

17. Verfahren nach Anspruch 16, dadurch gekennzeichnet, daß mit Hilfe des Systems $Ax=By=e$ die Gleichung für die über die betreffenden Abtastproben im Intervall gemittelte Fehlerenergie in eine Gleichung folgender Form umgesetzt wird:

$$e^Te=(y^TB^T+x^TA^T)(Ax+By)$$

worin der Vektor x_{min}, für den die Fehlerenergie minimal ist, durch folgende Formel gegeben wird:

$$(A^TA)x_{min}=-A^TBy$$

18. Verfahren nach Anspruch 17, dadurch gekennzeichnet, daß der Vektor x_{min} durch Auflösung des folgenden Systems berechnet wird:

$$Vx_{min}=w$$

— worin $V=A^TA$ ist, wobei $(A^TA)_{i,j}=\lambda_{t_i-t_j}$ ist,
worin i, j=1 ... m ist,
— worin $w=(A^TB)y$ mit

$$(A^TBy)_i=\sum_{k=0}^{N'-1}\lambda_{t_i-k}S_k \text{ met } i=1,\ldots, m$$

$k\neq t_1,\ldots t_m$
wobei $(A^TB)_{i,k}=\lambda_{t_i-k}$,
wobei $i=1\ldots m$
$k\neq t_1 \ldots t_m$ sind,

— und worin $\lambda_k=\sum_{l=0}^{p} a_l a_{l+k}$ ist, wobei

wobei $k=-p\ldots, 0,\ldots p$
$a_l=0$ ist, für $l>p$ und $l<0$ sind.

19. Verfahren nach Anspruch 18, dadurch gekennzeichnet, daß die unrichtigen Abtastproben zu aufeinander folgenden Zeitpunkten $t_1 \ldots t_m$ auftreten und das System $Vx_{min}=w$ mit Hilfe des Levinson-Algorithmus aufgelöst wird.

20. Verfahren nach einem oder mehreren der vorangehenden Ansprüche, dadurch gekennzeichnet, daß das Verfahren wenigstens einmal wiederholt wird, wobei die Bewertungsfaktoren der am besten geeigneten Rekusionsformel aus den Werten der richtungen Abtastproben im Intervall und den Werten der unrichtigen und mit Hilfe des Verfahrens abgeschätzten Abtastproben berechnet werden.

21. Anordnung zur Verwirklichung des Verfahrens nach Anspruch 1, dadurch gekennzeichnet, daß sie folgende Mittel enthält:

— Eingangsmittel zum Eingeben der Werte der Abtastproben des abgetasteten Signals,

— Detektormittel zum Detektieren unrichtiger Abtastproben,

— Mittel zum Ableiten eines ersten Abtastintervalls, in dem sich die Abtastproben mit fehlerhaften Werten befinden, aus der Anzahl von Abtastproben mit fehlerhaften Werten,

— Mittel zum Ableiten der Anzahl von Abtastproben, deren Werte in der Rekursionsformel gewogen werden, aus der Anzahl von Abtastproben mit fehlerhaften Werten,

— Mittel zum Bestimmen der Bewertungsfaktoren der Rekursionsformel aus de Rekursionsformeln mit unbekannten Bewertungsfaktoren für eine Anzahl von Abtastproben im ersten Abtastintervall unter Verwendung der Werte der Abtastproben in diesem Intervall, wobei eine erste Abschätzung für die Werte der Abtastproben mit fehlerhaften Werten aufgestellt wird,

— Mittel zum Bestimmen eines zweiten Abtastintervalls im ersten Abtastintervall, in dem sich die Abtastproben mit fehlerhaften Werten befinden, aus de Anzahl von Abtastproben, deren Werte in der Rekursionsformel gewogen sind,

— Mittel zum Ableiten abgeschätzter Werte für die Abtastproben mit fehlerhaften aus den Rekursionsformeln mit bekannten Bewertungsfaktoren für eine Anzahl von Abtastproben im zweiten Intervall unter Verwendung der Werte der richtigen Abtastproben in diesem Intervall, und

— Ausgangsmittel zum Ausgeben der Werte von Abtastproben des abgetasteten Signals.

22. Anordnung nach Anspruch 21, dadurch gekennzeichnet, daß die Mitteln zum Bestimmen der Bewertungsfaktoren folgende Elemente enthalten:

— Mitteln zum Berechnen der Autokorrelationskoeffizienten:

$$r(j)=\frac{1}{N}\sum_{k=0}^{N-j-1} S_k S_{k+j}$$

worin $j=(i-k)=0 \ldots p$ und $i, k=0 \ldots p$ sind,

— Mittel zum Auflösen des Systems:

$$Ra=b$$

— worin $r_{i,k}=r(j)$ ist,
wobei $j=(i-k)=0 \ldots p-1$,
$i, k=1 \ldots p$,
$b=-[r(1), r(2) \ldots r(p)]^T$
$a=[a_1 \ldots a_p]^T$,
worin a_i die Bewertungsfaktoren sind.

23. Anordnung nach Anspruch 22, dadurch gekennzeichnet, daß die Mittel zum Berechnen der Autokorrelationskoeffizienten Mittel zum Reduzieren des Quantisierungsgrades der Werte der Abtastproben enthalten.

24. Anordnung nach Anspruch 22, dadurch gekennzeichnet, daß die Mittel zum Berechnen der Autokorrelationskoeffizienten Mittel zum Bestimmen des Produkts der Werte von zwei Abtastproben mittels einer Tabelle mit möglichen Werten dieses Produkts enthalten.

25. Anordnung nach Anspruch 22, 23, oder 24, dadurch gekennzeichnet, daß die Mittel zum Auflösen des Systems $Ra=b$ den Levinson-Durbin-Algorithmus enthalten.

26. Anordnung nach einem oder mehreren der Ansprüche 21 bis 25, dadurch gekennzeichnet, daß sie folgende Mittel enthält:

— Mittel zum Berechnen der Anzahlen:

$$\lambda_k=\sum_{l=0}^{p} a_l a_{l+k},$$

worin $k=-p \ldots, 0, \ldots p$ und $a_l=0$ für $l>p$ und $l<0$ sind, und

— Mittel zum Berechnen der Anzahlen:

$$w_l=(A^T B y)_l=\sum_{k=0}^{N'-1} \lambda_{t_l-k} S_k \text{ ist,}$$

$$k \neq t_1, \ldots, t_m$$

worin $i=1 \ldots m$ und $k \neq t_1 \ldots t_m$ sind.

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27. Anordnung nach Anspruch 26, dadurch gekennzeichnet, daß sie weiter Mittel zum Auflösen des folgenden Systems enthält:

$$Vx_{min}=w$$

— worin $V=A^TA$ ist, wobei $(A^TA)_{i,j}=\lambda_{t_i-t_j}$ ist,
worin i, j=1 . . . m
— und worin $x_{min}=[S_{t_1} \ldots S_{t_m}]^T$ ist,
worin $S_{t_1} \ldots S_{t_m}$ die Werte der fehlenden Abtastproben sind.

28. Anordnung nach Anspruch 27, in der $S_{t_1} \ldots S_{t_m}$ aufeinanderfolgende Abtastproben sind, dadurch gekennzeichnet, daß die Mittel zum Auflösen des Systems $Vx_{min}=w$ einen Levinson-Algorithmus enthält.

29. Gerät zum Wiedergeben digitaler Information von einem Träger, insbesondere von einem optisch lesbaren Aufzeichnungsträger, auf dem digitale Information in Form optisch detektierbarer Gebiete abgewechselt mit Zwischengebieten gespeichert ist, dadurch gekennzeichnet, daß das Gerät mit einer Anordnung nach einem oder mehreren der vorangehenden Ansprüche 21 bis 28 versehen ist.

Revendications

1. Procédé pour la correction de valeurs erronées d'échantillons d'un signal echantilloné de façon équidistante à l'aide d'une formule de récurrence exprimant les échantillons erronés comme une somme pondérée des valeurs d'un nombre d'échantillons précédents, caractérisé par les étapes de
— la dérivation à partir du nombre d'échantillons présentant des valeurs erronées d'un premier intervalle d'échantillonnage dans lequel sont situés les échantillons présentant des valeurs erronées
— la dérivation à partir du nombre d'échantillons présentant des valeurs erronées du nombre d'échantillons dont les valeurs sont pondérées dans la formule de récurrence
— la détermination des coefficients de pondération de la formule de récurrence à partir des formules de récurrence présentant des coefficients de pondération inconnus pour un nombre d'échantillons dans le premier intervalle d'échantillonnage à l'aide des valeurs des échantillons dans cet intervalle, une première valeur estimée étant prise pour les valeurs des échantillons présentant des valeurs erronées,
— la détermination à partir du nombre d'échantillons dont les valeurs sont pondérées dans la formule de récurrence d'un second intervalle d'échantillonnage dans le premier intervalle d'échantillonnage dans lequel sont situés les échantillons présentant des valeurs erronées,
— la dérivation de valeurs estimées pour des échantillons présentant des valeurs erronées à partir des formules de récurrence présentant des coefficients de pondération connus pour un nombre d'échantillons dans le second intervalle à l'aide des valeurs des échantillons corrects dans cet intervalle.

2. Procédé selon la revendication 1, caractérisé en ce que pour déterminer les coefficients de pondération de la somme pondérée de la formule de récurrence, on effectue les étapes suivantes:
— l'établissement d'une formule de récurrence pour un nombre d'échantillons dans l'intervalle, formule qui contient une somme pondérée des valeurs d'un nombre d'échantillons précédents, la différence entre la valeur d'un échantillon et la valeur de la somme pondérée étant exprimée comme un terme d'erreur, une première valeur estimée étant prise pour les valeurs des échantillons erronés et le nombre d'échantillons pour lesquels est établie une formule de récurrence étant pratiquement égale à la différence entre le nombre d'échantillons dans l'intervalle et le nombre d'échantillons dont les valeurs sont pondérées dans la somme pondérée.
— détermination de l'énergie d'erreur correspondant aux termes d'erreur et constituant la moyenne des échantillons dans l'intervalle et
— minimiser l'énergie d'erreur en fonction de chacun des coefficients de pondération.

3. Procédé selon la revendication 1 ou 2, caractérisé en ce que pour déterminer le coefficient de pondération de la somme pondérée de la formule de récurrence, le système d'équation à résoudre de la forme: voir la formule (20), est remplacé par un système de Toeplitz de la formule: voir la formule (21).

4. Procédé selon la revendication 3, caractérisé en ce que les estimateurs pour le coefficient d'autocorrelation sont donnés par:
voir la formule (7).

5. Procédé selon la revendication 3 ou 4, caractérisé en ce que pour calculer les estimateurs pour les coefficients d'autocorrélation, les valeurs des échantillons sont décrites à un degré de quantisation inférieur au degré de quantisation réel des échantillons.

6. Procédé selon la revendication 4 ou 5, caractérisé en ce que pour calculer les estimateurs des coefficients d'autocorrélation, le résultat d'une multiplication des valeurs de deux échantillons est cherché dans un tableau contenant les résultats possibles de la multiplication.

7. Procédé selon la revendication 3, 4, 5 ou 6, caractérisé en ce que le système Teoplitz $R_2a=b_2$ est résolu à l'aide de l'algorithme de Levinson-Durbin.

8. Procédé selon la revendication 1, caractérisé en ce qu'une valeur préalablement déterminée est adoptée pour le nombre maximal p d'échantillons dans la formule de récurrence.

9. Procédé selon la revendication 1, caractérisé en ce que le nombre maximal p d'échantillons de la récurrence dépend du nombre m d'échantillons erronés.

10. Procédé selon la revendication 9, caractérisé en ce que le nombre maximal p d'échantillons augmente suivant une fonction linéaire du nombre d'échantillons erronés.

11. Procédé selon la revendication 10, caractérisé en ce que le signal est un signal audio qui est échantilloné à une fréquence d'environ 44,1 kHz et la relation entre le nombre maximal p d'échantillons de la récurrence et le nombre m d'échantillons erronés est donné par $p \approx 3m+2$.

12. Procédé selon la revendication 1, caractérisé en ce que le nombre N d'échantillons dans l'intervalle est déterminé par le nombre m d'échantillons erronés.

13. Procédé selon la revendication 12, caractérisé en ce que le nombre N d'échantillons augmente suivant une fonction linéaire du nombre m d'échantillons erronés.

14. Procédé selon la revendication 10, caractérisé en ce que le signal est un signal audio qui est échantilloné à une fréquence d'environ 44,1 kHz et la relation entre le nombre N d'échantillons dans l'intervalle et le nombre d'échantillons erronés est donnée par $N \approx 32m$.

15. Procédé selon l'une des revendications précédentes, caractérisé en ce que pour estimer les valeurs des échantillons erronés, on effectue les étapes suivantes:

— dans le premier intervalle dans lequel sont situés les échantillons erronés est déterminé un deuxième intervalle dont le premier échantillon est situé au moins nombre d'échantillons égal au nombre d'échantillons dans la formule de récurrence avant le premier échantillon erroné et dont le dernier échantillon est situé au moins un nombre d'échantillons égal au nombre d'échantillons dans la formule de récurrence après le dernier échantillon erroné,

— pour au moins les échantillons à partir du premier échantillon erroné jusqu'à et y compris le dernier échantillon dans le deuxième intervalle, la formule de récurrence est établie avec les coefficients de pondération calculés,

— l'énergie d'erreur constituant la moyenne des échantillons en question dans l'intervalle et correspondant aux termes d'erreur des formules de récurrence est déterminée et,

— les valeurs estimées des échantillons erronés sont déterminées par minimiser l'énergie d'erreur suivant une fonction des valeurs des échantillons eronés.

16. Procédé selon la revendication 15, caractérisé en ce que les formules de récurrence constitue le système d'équations de récurrence de la forme: voir la formule (10) et ce système est transformé en un système de la forme voir la formule (12).

17. Procédé selon la revendication 16, caractérisé en ce que l'équation pour l'énergie d'erreur constituant la moyenne des échantillons en question dans l'intervalle est transformée, à l'aide du système $Ax+By=e$, en une équation de la formule

voir la formule (13),

dans laquelle le vecteur x_{min} pour lequel l'énergie d'erreur est minimale est donné par

voir la formule (14).

18. Procédé selon la revendication 17, caractérisé en ce que le vecteur x_{min} est calculé par résolution du système:

voir la formule (15).

19. Procédé selon la revendication 18, caractérisé en ce que les échantillons erronés sont situés à des moments successifs $t_1, \ldots t_m$ et le système $Vx_{min}=w$ est résolu à l'aide de l'algorithme de Levinson.

20. Procédé selon l'une des revendications précédentes, caractérisé en ce que le procédé est répété au moins une fois, les coefficients de pondération de la formule de récurrence s'adaptant le mieux étant calculés à partir des valeurs des échantillons corrects dans l'intervalle et les valeurs des échantillons erronés sont estimées à l'aide de ce procédé.

21. Dispositif pour la mise en oeuvre du procédé selon la revendication 1, caractérisé en ce que le dispositif comprend

— des moyens d'entrée pour l'entrée des valeurs d'échantillons du signal échantilloné,

— des moyens de détection pour la détection d'échantillons erronés,

— des moyens pour dériver à partir du nombre d'échantillons présentant des valeurs erronées d'un premier intervalle d'échantillonnage dans lequel sont situés les échantillons présentant des valeurs erronées

— des moyens pour dériver à partir du nombre d'échantillons présentant des valeurs erronées le nombre d'échantillons dont les valeurs sont pondérées dans la formule de récurrence

— des moyens pour déterminer les coefficients de pondération de la formule de récurrence à partir des formules de récurrence présentant des coefficients de pondération inconnus pour un nombre d'échantillons dans le premier intervalle d'échantillonnage à l'aide des valeurs des échantillons dans cet intervalle, une première valeur estimée étant prise pour la valeur des échantillons présentant des valeurs erronées,

— des moyens pour déterminer à partir du nombre d'échantillons dont les valeurs sont pondérées dans la formule de récurrence un second intervalle d'échantillonnage dans le premier intervalle d'échantillonnage dans lequel sont situés les échantillons présentant des valeurs erronées,

— des moyens pour dériver des valeurs estimées pour les échantillons présentant des valeurs erronées à partir des formules de récurrence présentant des coefficients de pondération inconnus pour un

nombre d'échantillons dans le deuxième intervalle à l'aide des valeurs des échantillons corrects dans cet intervalle et

— des moyens de sortie pour la sortie des valeurs d'échantillons du signal échantilloné.

22. Dispositif selon la revendication 21, caractérisé en ce que les moyens pour déterminer les coefficients de pondération sont munis de

— moyens pour calculer des coefficients d'autocorrélation:
voir la formule (7) et

— moyens pour résoudre le système
voir la formule (16).

23. Dispositif selon la revendication 22, caractérisé en ce que les moyens pour calculer les coefficients d'autocorrélation contiennent des moyens pour réduire le degré de quantisation des valeurs des échantillons.

24. Dispositif selon la revendication 22, caractérisé en ce que les moyens pour calculer les coefficients d'autocorrélation contiennent des moyens pour déterminer le produit des valeurs des deux échantillons à l'aide d'un tableau de valeurs possibles de ce produit.

25. Dispositif selon la revendication 22, 23 ou 24, caractérisé en ce que les moyens pour résoudre le système $Ra=b$ contiennent l'algorithme de Levison-Durbin.

26. Dispositif selon l'une des revendications 21 à 25, caractérisé en ce qu'il comprend:

— des moyens pour calculer les nombres:
voir la formule (17) et

— des moyens pour calculer les nombres
voir la formule (18).

27. Dispositif selon la revendication 26, caractérisé en ce qu'il comprend en outre des moyens pour résoudre le système
voir la formule (19).

28. Dispositif selon la revendication 27, $S_{t_1} \ldots S_{t_m}$ étant des échantillons successifs, caractérisé en ce que les moyens pour résoudre le système $Vx_{min}=w$ contiennent un algorithme de Levinson.

29. Appareil pour reproduire de l'information numérique à partir d'un porteur, notamment un porteur d'enregistrement lisible par voie optique, dans lequel de l'information numérique est stockée sous forme de zones pouvant être détectées par voie optique et alternant avec des zones intermédiaires, caractérisé en ce que l'appareil est muni d'un dispositif selon l'une des revendications précédentes 21 à 28.

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