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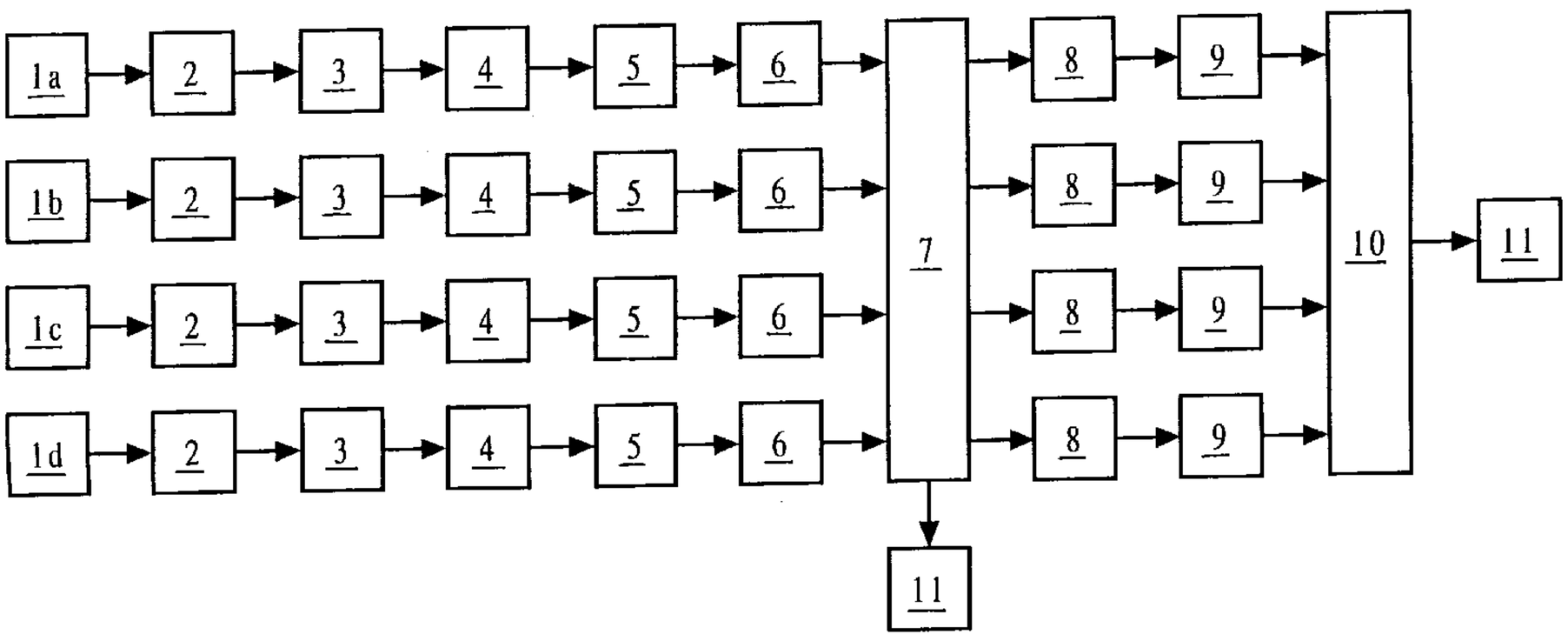
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(54) **PROCEDURE DE MESURE POUR CORRIGER AUDITIVEMENT L'ESTIMATION DE LA QUALITE DE SIGNAUX AUDIO**

(54) **MEASUREMENT PROCEDURE FOR AURALLY CORRECT QUALITY ASSESSMENT OF AUDIO SIGNALS**



(57) The measurement procedure assesses the interference of an audio signal or test signal, respectively, (1a,b) by comparing it with a reference signal (1c,d) that is not subject to interferences. After pre-filtering (2) with the transmission functions of the outer and middle ear, the input signals are converted into a time-tonality representation. The squares of sums (5) of the filter output signals are calculated (rectification) and folding of the filter outputs is carried out with a smearing function (4). The folding can be done either before or after rectification. Level differences between test signals and reference signals, as well as linear distortions of the reference signals are compensated at (7) and evaluated separately. Then, at (8), a frequency-dependent offset is added in order to model the inherent noise of the auditory process., and the output signals are time smeared at (9). In order to save computer time, some of this time smearing can take place immediately after rectification (4). After time smearing (8) (low-pass filtering) it is then permissible to underscan the signals. A series of output values can be calculated at (10) by comparison of the resulting aurally corrected time-frequency patterns of test and reference signals (1a,b and 1c,d); these output values provide a assessment of the perceivable interference.

Abstract

The measurement procedure assesses the interference of an audio signal or test signal, respectively, (1a,b) by comparing it with a reference signal (1c,d) that is not subject to interferences. After pre-filtering (2) with the transmission functions of the outer and middle ear, the input signals are converted into a time-tonality representation . The squares of sums (5) of the filter output signals are calculated (rectification) and folding of the filter outputs is carried out with a smearing function (4). The folding can be done either before or after rectification. Level differences between test signals and reference signals, as well as linear distortions of the reference signals are compensated at (7) and evaluated separately. Then, at (8), a frequency-dependent offset is added in order to model the inherent noise of the auditory process., and the output signals are time smeared at (9). In order to save computer time, some of this time smearing can take place immediately after rectification (4). After time smearing (8) (low-pass filtering) it is then permissible to underscan the signals. A series of output values can be calculated at (10) by comparison of the resulting aurally corrected time-frequency patterns of test and reference signals (1a,b and 1c,d); these output values provide a assessment of the perceivable interference.

Measurement Procedure for Aurally Correct Quality Assessment of Audio Signals

The present invention relates to a measurement procedure for the aurally correct quality assessment of audio signals as set out in the preamble to Patent Claim 1.

Measurement procedures for aurally correct quality assessment of audio signals are known in principle. The fundamental structure of such a measurement procedure includes mapping the input signals on an aurally corrected time-frequency mapping, a comparison of this representation, and calculation of the individual numerical values for the assessment of audible interference. To this end, reference is made to the following publications:

Schroeder, M.R.; Atal, B.S.; Hall, J.L.: Optimizing digital speech coders by exploiting masking properties of the human ear. J. Acoustic Soc. Am., Vol. 66 (1979), No. 6, December, pp. 1647 - 1652.

Beerends, J.G.; Stemerdink, J.A.: A Perceptual Audio Quality Measure Based on a Psycho-acoustic Sound Representation. J. AES, Vol. 40 (1992), No. 12, December, pp. 963 - 978.

Brandenburg, K.H.; Sporer, Th.: NMR and Masking Flag: Evaluation of Quality Using Perceptual Criteria. Proceedings of the AES

11th International Conference, Portland, Oregon, USA, 1992, pp. 169 - 179.

As can be learned from these papers, the models used for evaluating coded audio signals use FFT algorithms and for this reason they have to be converted from the linear frequency division laid down by FFT to an aurally correct frequency division. Because of this, time resolution is sub-optimal. In addition, folding is effected with a smearing function after rectification or summation.

For this reason, it is the task of the present invention to create an objective measurement procedure for aurally correct quality assessment of audio signals by using new, faster algorithms for designing linear-phase filters, the run time of the audible interference being calculated taking into account the time change of the envelope curve at the individual filter outlets, a matched filter bank being meant for use, whereby an optimal time resolution is to be achieved, together with a significant saving of computer time vis-à-vis other filter banks.

The solution to this task as proposed by the present invention is set out in Patent Claim 5 or Patent Claim 7, respectively.

Other solutions or configurations of the present invention are set out in Patent Claim 2 to Patent Claim 23.

An important advantage of the process according to the present invention is that a precise acoustic model is achieved, since audible interference has been calculated, taking the time change at the individual filter outputs into consideration.

In addition, an aurally matched filter bank is used, whereby an optimal time resolution is achieved, and the time behaviour of the filter (impulse response, etc.) corresponds directly to the level dependency of the transmission functions. The phase information in the filter channels is retained. As already stated, in the solutions known up to the present time, the folding with the associated smearing function is first effected after rectification or sum formation. A signal dependency of the filter characteristics is achieved in that the filter outputs are folded in the frequency range with a level dependent smearing function, before rectification/sum formation.

The fact that a new, faster algorithm for recursive computation of linear-phase filters is used results in a significant saving of computer time, simple development, and filters that are more easily varied than formerly used conventional, recursive filters.

Signal components present in the original signal, which are modified only with respect to their spectral distribution, are separated from additive interference or interference generated by non-linearities, such separation being effected by

analysis of the orthogonality relationship between the time curves of the envelope curves at corresponding filter outputs of the signal to be analyzed and of the original signal. The separation of these interference components correspond better to the actual auditory impression.

The filter-bank algorithm is realised in the following way:

- an undamped sinus oscillation with the desired filter mid-frequency is generated from each incoming pulse by recursive complex multiplication.
- The sinus oscillation that is associated with an input pulse is truncated again by subtraction of the input pulse, delayed by the time corresponding to the reciprocal value of the desired filter band width, and multiplied by the phase angle corresponding to the delay.
- A damping curve corresponding to the Fourier transform of a cos^(n-1) shaped time window is generated by folding in the frequency range, by weighted summing of n filter outputs each, of identical bandwidth, and by a mid-frequency, offset in each instance by one period, of the sin(x)/x shaped damping curve resulting from Step 2. By so doing, it is possible to form the damping curve in the immediate area of the filter mid-frequencies, and arrive at a sufficiently high non-pass attenuation.

Additional advantages, features, and potential applications are set out in the following description, which is based on embodiments show in the drawings appended hereto.

The present invention will be described in greater detail below, on the basis of the drawings appended hereto. The terms and associated reference numbers used in the list of reference numbers appended at the end hereof are used in the description, the patent claims, in the abstract, and in the drawings. These drawings show the following:

Figure 1: A structure of the measurement procedure;

Figure 2: A filter structure.

The present measurement procedure analyzes the interference of an audio signal by comparison with a reference signal that is not subjected to interference. After filtering with the transmission functions of the outer and middle ear, the input signals are transformed into a time-tonality representation by an auditory -matched filter bank. The squares of the totals of the filter output signals are computed (rectification) and the filter outputs are folded with a smearing function. In contrast to the procedure known up to the present, the folding can be effected either before or after rectification. Differences in the levels of the test signal and the reference signal, as well as linear distortion in the test signal, are compensated and

analyzed separately. An offset that is a function of frequency is then added in order to model the inherent hearing [auditory] noise, and the output signals are time smeared. Some of this time smearing can be effected directly after rectification, in order to save computer time. Under-scanning of the signals is permissible after time smearing (low-pass filtering). A series of output values can be calculated by comparison of the resulting, aurally-corrected time-frequency patterns of the test and reference signals; these output values then provide an estimate of the perceptible interference.

First, the structure or design of the measurement procedure that is shown in Figure 1 as an exemplary embodiment will be explained. The test signals 1a, 1b for the left or right channel, respectively, and the reference signals 1c, 1d for the left or right channel, respectively, are passed to prefilter 2 for prefiltering. After prefiltering, the actual filtering is carried out in the filter bank 3. Subsequently, spectral smearing 4 and calculation of the squares of the totals 5 are carried out. The boxes 6 in the drawings are a symbolic representation of the time smearing. This is followed by level- and frequency response compensation 7, this also providing the output parameter 11. After level- and frequency response compensation, inherent noise is totalled at 8, and then time smearing is completed at 9.

In the structure that is shown, calculation of output parameters 11 is effected at the symbolically represented block 11. Level- and frequency-response compensation 7 can also be completed between operations 9 and 10.

First, calculation of the excitation pattern by the aurally matched filter bank 3 will be described.

The filter bank 3 consists of a selectable number of filter pairs for test and reference signals 1a, b, or 1d, c (values between 30 and 200 are appropriate). The filters can be evenly distributed as desired over almost any tone-pitch scales. A suitable tone-pitch scale is the following one, as proposed by Schroeder:

$$z / Bark = 7 \cdot \operatorname{arsinh}\left(\frac{f / Hz}{650}\right) \qquad \text{Equation 1}$$

The filters are linear-phase filters and are defined by impulse responses of the following form:

$$h_{re}(t) = \cos^{n}(\pi \cdot bw \cdot t) \cdot \cos(2\pi \cdot f_c \cdot t) \Bigg|_{|t| < \frac{1}{2 \cdot bw}} \qquad \text{Equation 2}$$

and

$$h_{im}(t) = \cos^n(\pi \cdot bw \cdot t) \cdot \sin(2\pi \cdot f_c \cdot t) \Big|_{|t| < \frac{1}{2 \cdot bw}} \qquad \text{Equation 3}$$

The value n defined the non-pass attenuation; it should be ≥ 2.

In order to account for simultaneous coverage, the output values of the filter bank 3 are spectrally smeared with 31 dB/Bark on the lower side, and between -24 and -6 dB/Bark on the upper side, which is to say, that cross-talk is generated between the filter outputs. The upper side is calculated as a function of level as follows:

Equation 4

$$s = \min\left(-6\frac{dB}{Bark}, -24\frac{dB}{Bark} + 0.2Bark^{-1} \cdot L / dB\right)$$

The level L is calculated independently for each filter output, from the squared sum 5 of the corresponding output value, low-pass filtered with a time constant of 10 ms. This smearing is effected independently for the filters that represent the real portion of the signal (Equation 2) and the filters that represent the imaginary portion of the signal (Equation 3). As an alternative, the level can also be calculated without any low-pass filtering, and in place of this the factor that determines cross-talk, which results for the anti-logarithm of the steepness of the side (Equation 4) can be low-pass filtered. Since this

folding operation is quasi-linear, and thus preserves the relationship between the resulting frequency response and the resulting impulse response, it can be taken regarded as a part of filter bank 3.

Since filter bank 3 delivers pairs of output signals that are phase-shifted through 90°, rectification can be effected by forming the squared sum 5 of the filter outputs:

$$E(f_c,t) = A_m^2(f_c,t) + A_{lm}^2(f_c,t) \qquad \text{Equation 5}$$

Time smearing of the filter output signals is effected in two stages. In the first stage, the signals are averaged by a $\cos^2$ shaped time window, so that primarily the pre-coverage is modelled. Then, in the second stage, the post-coverage is modelled; this is described more precisely below. The $\cos^2$ shaped time window has a length of 400 scanning values at a maximum scanning rage of 48 kHz. The distance between the maximum of the time window and its 3 dB point thus amounts to some 100 scanning values, or 2 ms, that is in keeping with a time span that is frequently accepted for the pre-coverage.

Level differences and linear distortion (frequency response of the test object) between test and reference signal 1a,b or 1c,d, respectively, can be compensated and thus separated from the assessment [the German used here: Bewerung, is

meaningless; Bewertung (assessment, evaluation) seems logical to this layman!) of other kinds of interference.

For level compensation, the current squared sum at the filter outputs are smoothed over time by low passes of the first order. The time constants that are used are selected as a function of the mid-frequency of the particular filter:

$$\tau = \tau_0 + \frac{100Hz}{f_c} \cdot (\tau_{100} - \tau_0) \quad \left| \begin{array}{lcl} \tau_{100} & = & 0{,}004\text{–}1s \\ \tau_0 & = & 0{,}004\text{–}1s, \textit{ wobei} \end{array} \right. \qquad \text{Equation 6}$$

$\tau_{100} \geq \tau_0$.

A correction factor $corr_{total}$ is calculated from the filter output values P_{test} and P_{ref}:

$$corr_{total} = \left(\frac{\sum \sqrt{P_{Test} \cdot P_{Ref}}}{\sum P_{Test}} \right)^2 \qquad \text{Equation 7}$$

If this correction factor is greater than 1, the reference signal 1a; b is divided by the correction factor, otherwise the test signal 1c; d is multiplied by the correction factor.

Correction factors are calculated for each filter channel from the orthogonality relationship between the time envelope curves of the filter outputs of test and reference signals 1a, b; 1c, d for each filter channel:

$$ratio_{f,t} = \frac{\int_{-\infty}^{0} e^{\frac{t}{\tau}} \cdot X_{Test} \cdot X_{Ref} dt}{\int_{-\infty}^{0} e^{\frac{t}{\tau}} \cdot X_{Ref} \cdot X_{Ref} dt}$$ Equation 8

The time constants are determined by Equation 6. If $ratio_{f,t}$ is greater than 1, the correction factor for the test signal is set to $ratio_{f,t}^{-1}$ and the correction factor for the reference signal is set to 1. In the opposite case, the correction factor for the reference signal is set to $ratio_{f,t}$, and the correction factor for the test signal is set to 1.

The correction factors are time smoothed across a plurality of adjacent filter channels and with the same time constants, as described above.

A frequency-dependent offset for modelling the inherent noise of the auditory process is added to the squared total at all filter outputs. An additional offset for taking background noise into account can similarly be added (but in the normal case it will be set to 0).

$$E(f_c,t) = E(f,t) + 10^{0.364\left(\frac{f_c}{kHz}\right)^{-0.8}}$$ Equation 9

The current squared total in each filter channel is time smeared by a low pass of the first order, with a time constant of approximately 10 ms, in order to model the post-coverage. If desired, the time constants can be calculated as a function of the mid-frequency of the particular filter. In this

case, it is at 50 ms for lower frequencies, and at 8 ms for higher requencies (as in Equation 6).

Prior to the above-described second stage of time smearing, a simple approximation for loudness is calculated, in that the squared totals at the filter outputs are taken to be at most 0.3. This value $\bar{E}$ and the sum of its time derivative $d\bar{E}/dt$ are smoothed with the same constants as described above. A measure for the envelope-curve modulation in each channel is determined from the results of the time smoothing $\bar{E}_{der}$:

$$\mathbf{mod}(f_c,t)=\frac{\overline{E}_{der}(f_c,t)}{1+\overline{E}(f_c,t)} \qquad \text{Equation 10}$$

The most important output parameter, and the one that is most highly correlated with subjective hearing-test data, is the loudness of the interference during choking by the information signal. The input values for this are the squared totals in each filter channel E_{ref} and E_{test} ("excitation"), the envelope-curve modulation, the inherent noise of the auditory process ("base excitation") E_{HS}, and the constants E_0 and α. The choked interference loudness is calculated according to

$$NL(f_c,t)=\left(\frac{1}{s_{test}}\cdot\frac{E_{HS}}{E_0}\right)^{0.23}\cdot\left[\left(1+\frac{\max\left(s_{test}\cdot E_{test}-s_{ref}\cdot E_{ref},0\right)}{E_{HS}+s_{ref}\cdot E_{ref}\cdot\beta}\right)^{0.23}-1\right] \qquad \text{Equation 11}$$

wherein

$$E_{HS} = 10^{0.364\left(\frac{f_c}{kHz}\right)^{-0.8}}$$

$$E_0 = 10^{4}$$

$$\alpha = 1.0$$

$$s = 0.04 \cdot \mathrm{mod}(f_c, t) / Hz + 1$$

Equation 11 is so constructed that it provides the specific loudness of the interference if there is no masker, and provides it in the approximate ratio between interference and masker if the interference is very small in relation to the masker. The factor β, which determines the choking, is calculated according to the following equation:

$$\beta = \exp\left(-\alpha \cdot \frac{E_{test} - E_{ref}}{E_{ref}}\right) \qquad \text{Equation 12}$$

The "choked interference loudness" corresponds to the middle value of this amplitude over time and filter channels. In order to determine linear distortion, this same calculation is completed once more without frequency-response compensation, with test and reference signals being exchanged in the above equation. The resulting output parameter is designated "loudness-deficient signal parts." A well-founded prediction of the subjectively perceived signal quality of a coded audio signal is possible given these two output values. As an alternative, linear distortion can also be determined, with the reference signal

before signal compensation being used as the test signal. An additional output value is the modulation differential that results from standard-standardization [normalization, scaling] of the difference of the modulation of test and reference signal on the modulation of the reference signal. When this is done, an offset is added to the reference signal during normalization in order to limit the calculated values in the case of very small modulation of the reference signal:

$$\text{modulation differential} = \frac{\text{modtest - modref}}{\text{offset + modref}}$$

The modulation difference is averaged over time and filter bands.

The modulation used on the input side results from normalization of the time derivative of the current values on their time-smoothed value.

Figure 2 shows a filter structure for recursive calculation of a simple bandpass with finite impulse response (FIR).

The signal is processed separately for real part (upper path) and imaginary part (lower path). Since the input signal X is originally purely real, initially there is no lower path. The input signal X is delayed by N sampling values (1) and after multiplication by a complex-value factor $\cos(N.\phi) + j.\sin(N.\phi)$ it is subtracted from the original input signal (2). The resulting

signal V is added to output signal delayed by one scan value (3). The result, multiplied by an additional complex-value factor $\cos(\phi)+j.\sin(\phi)$ provides the new output signal Y (4). The over-scored designator for V and Y each mark the imaginary part.

The second complex multiplication continues the input signal periodically. The addition of the delayed input signal weighted by the first complex multiplication interrupts continuation of the input signal once again after N scanning values.

The total filter, consisting of real-part and imaginary-part outputs has the amplitude frequency response:

$$A(f) = N \cdot \frac{si\left(\frac{N}{2}\left(\varphi - \frac{2 \cdot \pi \cdot f}{f_A}\right)\right)}{si\left(\frac{1}{2}\left(\varphi - \frac{2 \cdot \pi \cdot f}{f_A}\right)\right)},$$

wherein f_A indicates the scanning frequency.

The non-pass attenuation of these band-pass filters, which is initially low, can be increased if one calculates K+1 of such band-pass filters with equal impulse response lengths N, but different values of ϕ in parallel, matches their phase responses to each other by an additional complex multiplication, and totals their output signal, weighted:

$$A(f)=\sum_{k=0}^{K} w_k \cdot A_k(f)$$

with

$$\varphi_k=\frac{2\cdot\pi\cdot f_M}{f_\Lambda}+\left(k-\frac{K}{2}\right)\cdot\frac{2\pi}{N}$$

(f_M: mid-frequencies of the band pass) and

$$w_k=\frac{2\pi}{N}\cdot 2^{-K}\cdot\binom{K}{k}$$

The non-pass attenuation of the resulting filter decreases with the (K+1)-th power of the distance of the signal frequency to the mid-frequency of the filter. The impulse response of the total filter has the form

$$a_K(n)=\sin^K\left(\frac{\pi}{N}n\right)\cdot\cos\left(\frac{2\cdot\pi\cdot f_M}{f_\Lambda}\cdot n\right) \quad \Big| \quad 0\le n<N$$

for the real part, and

$$a_K(n) = \sin^K\left(\frac{\pi}{N}n\right) \cdot \sin\left(\frac{2 \cdot \pi \cdot f_M}{f_\Lambda} \cdot n\right) \quad \Big| \quad 0 \le n < N$$

for the imaginary part. This corresponds to the characteristics described in Equation 2 and Equation 3.

Reference Numbers:

1a	Test signal, left-hand channel
1b	Test signal, right-hand channel
1c	Reference signal, left-hand channel
1d	Reference signal, right-hand channel
2	Pre-filtering
3	Filter bank
4	Spectral smearing
5	Calculation of squared total
6	Time smearing
7	Level and frequency compensation
8	Addition on inherent noise
9	Time smearing
10	Calculation of output parameters
11	Output parameters

THE EMBODIMENTS OF THE INVENTION IN WHICH AN EXCLUSIVE PROPERTY OR PRIVILEGE IS CLAIMED ARE DEFINED AS FOLLOWS:

1. Measurement procedure for aurally corrected quality assessment of audio signals with the help of filters, time smearing, and level and frequency compensation, characterized in that the audio signal that is to be assessed is compared as a test signal (1a,b) to an original signal that is provided as a reference signal (1c,d); in that both signals or pairs of signals (1a,b; 1c,d) are broken down into the frequency range after pre-filtering (2) through a filter bank (3); in that an aurally corrected representation of the audio signals to be assessed as test signals (1a,b) is generated by the characteristics of the filter band (3) and subsequent time smearing (9) of the filter output signals; in that an assessment of the auditory impression that is to be expected is provided by comparison of the aurally corrected representations of test signal (1a,b) and reference signal (1c,d) after non-linear reshaping.

2. Procedure as defined in Patent Claim 1, characterized in that the filter bank (3) is aurally matched, and an undamped sinus oscillation with the desired filter mid-frequency is

generated from each incoming signal by recursive complex multiplication; and in that the sinus oscillation that is associated with a test signal (1a,b) is truncated again by subtraction of the test signal (1a,b), delayed by the time corresponding to the reciprocal value of the desired filter band width, and multiplied by the phase angle corresponding to the delay.

3. Procedure as defined in Patent Claim 1, characterized in that an appropriate damping curve corresponding to the Fourier transform of a $\cos^n(n-1)$ shaped time window is generated from each n filter outputs of equal bandwidth and a mid-frequency offset in each instance by the reciprocal of the window length, by folding in the frequency range.

4. Procedure as defined in one of the Patent Claims 2 or 3, characterized in that at a greater distance from the filter mid-frequency, the damping curve can be determined by additional folding in the frequency range, in the transition between pass band and non-pass band.

5. Procedure as defined in the preamble to Patent Claim 1, characterized in that an undamped sinus oscillation with the desired filter mid-frequency is generated from each incoming

test signal (1a,b) by recursive complex multiplication; in that the sinus oscillation that is associated with a test signal (1a,b) is truncated by subtraction of the test signal (1a,b), delayed by the time corresponding to the reciprocal value of the desired filter band width and multiplied by the phase angle corresponding to the delay; in that an appropriate damping curve corresponding to the Fourier transform of a $\cos^n(n-1)$ shaped time window is generated from each n filter outputs of equal bandwidth and a mid-frequency offset in each instance by the reciprocal of the window length, by folding in the frequency range; in that at a greater distance from the filter mid-frequency the damping curve can be determined by additional folding in the frequency range (transition between pass band and non-pass band).

6. Procedure as defined in one of the Patent Claims 1 to 5, characterized in that the input test signals (1a,b) and the reference signals (1c,d) are each introduced as input values for a left-hand and a right-hand channel, which is to say, in pairs.

7. Procedure as defined in the preamble to Patent Claim 1, characterized in that the test signals (1a,b) and the

reference signals (1c,d) are first subjected to pre-filtering (2), subsequently passed to a filter bank (3); in that, subsequently, spectral smearing is effected; and then the square of the sum (5) is arrived at, after which time smearing is completed; in that the output values so obtained are subjected to level and frequency compensation (7); and in that, subsequently, addition of inherent noised is effected, whereafter once again time smearing (9) and the calculation (10) of output parameters (11) is effected, or step (7) is completed between step (9) and step (10).

8. Procedure as defined in one of the Patent Claims 1 to 7, characterized in that conversion to a time-tone representation is carried out after filtering with transmission functions of outer and middle ear input signals by an aurally-matched filter bank (3); in that, subsequently, the squared sums (5) of the filter output signals are calculated and a folding of the filter output signals is carried out with a smearing function (6).

9. Procedure as defined in Patent Claim 9, characterized in that the folding is effected before or after rectification.

10. Procedure as defined in one of the Patent Claims 1 to 9, characterized in that the level differentials between test and reference signal (1a, b; 1c,d) as well as linear distortion of the reference signal (1c,d) are compensated and evaluated separately.

11. Procedure as defined in one of the Patent Claims 1 to 9, characterized in that part of the time smearing is effected immediately after rectification.

12. Procedure as defined in one of the patent Claims 1 or 5, characterized in that an aurally-matched filter bank (3) is used, this achieving a signal dependency of the filter characteristics in that the filter outputs are folded prior to rectification/total formation with a level-dependent smearing function in the frequency range.

13. Procedure as defined in one of the Patent Claims 1 to 12, characterized in that signal parts that are present in the reference signal (1c,d) and only modified with respect to their spectral distribution, are separated from additive interference or from interference generated by non-linearities; and in that the separation of these interference parts is effected by evaluation of the

orthogonality relationship between the time progressions of the envelope curves at filter outputs of the test signal (1a,b) and of the reference signal (1c,d) that correspond to each other.

14. Procedure as defined in one of the Patent Claims 1 to 13, characterized in that the filter bank (3) comprises any selected number of filter pairs for test and reference signals (1a,b; 1c,d); and in that the filters are distributed evenly on largely any tone-level scales.

15. Procedure as defined in one of the Patent Claims 1 to 14, characterized in that

the output values of the filter bank (3) are spectrally smeared on the upper slope in order to take simultaneous coverage into account;

the level (L) is calculated as a function of each filter output from the squared total (5) of the corresponding output value, which has been low-pass filtered with a time constant, or is determined without a low-pass filter, and is place of this the smearing factor is low-pass filtered, and the smearing is completed independently for the filters that represent the real part of the signal and the filters that represent the imaginary part of the signal.

16. Procedure as defined in one of the Patent Claims 1 to 15, characterized in that
the time smearing of the filter output signals is performed in two stages, the signal being ascertained in the first stage by a cosineyy2-shaped time window, and a post-coverage being modulated in the second stage.

17. Procedure as defined in Patent Claim 16, characterized in that the cosineyy2-shaped time windows are between 1 - 16 ms long.

18. Procedure as defined in one of the Patent Claims 1 to 17, characterized in that the current squared totals (5) are time smoothed at the filter outputs by first-order low passes for purposes of level compensation;
the time constants that are used are selected as a function of the mid-frequency of the particular filter, and
a correction factor is calculated from the orthogonality relationship between spectral envelope curves of the time smoothed filter outlets of test and reference signals (1a,b; 1c,d).

19. Procedure as defined in Patent Claim 18, characterized in that

the test signal is multiplied by the correction factor if the correction factor is <1 and the reference signal is divided by the correction factor if the correction factor is >1.

20. Procedure as defined in one of the Patent Claims 1 to 19, characterized in that
correction factors are calculated from the orthogonality relationship that exists between the time envelope curves of the filter outputs of test and reference signals (1a,b; 1c,d).

21. Procedure as defined in Patent Claim 1, characterized in that a modulation differential is determined from the normalized (absolute) difference of the envelope curves of the test and reference signals for each filter channel, and each filter band, this being suitable--after time and spectral averaging --for evaluating specific audible interference.

22. Procedure as defined in one of the Patent Claims 1 to 21, characterized in that from input values in the form of the squares of sums (5) in each filter channel, envelope curve modulation, the inherent noise of the hearing process, and

constants, a choked interference loudness is determined and averaged over time and filter channels.

23. Procedure as defined in one of the Patent Claims 1 to 22, characterized in that

the input signal (X) is delayed by N scanning values and after multiplication by a complex factor is subtracted from the original input signal;

the resulting signal)V) is added to the output signal that is delayed by a scanning value;

the resultant, multiplied by a further complex-value factor, results in the new output signal.

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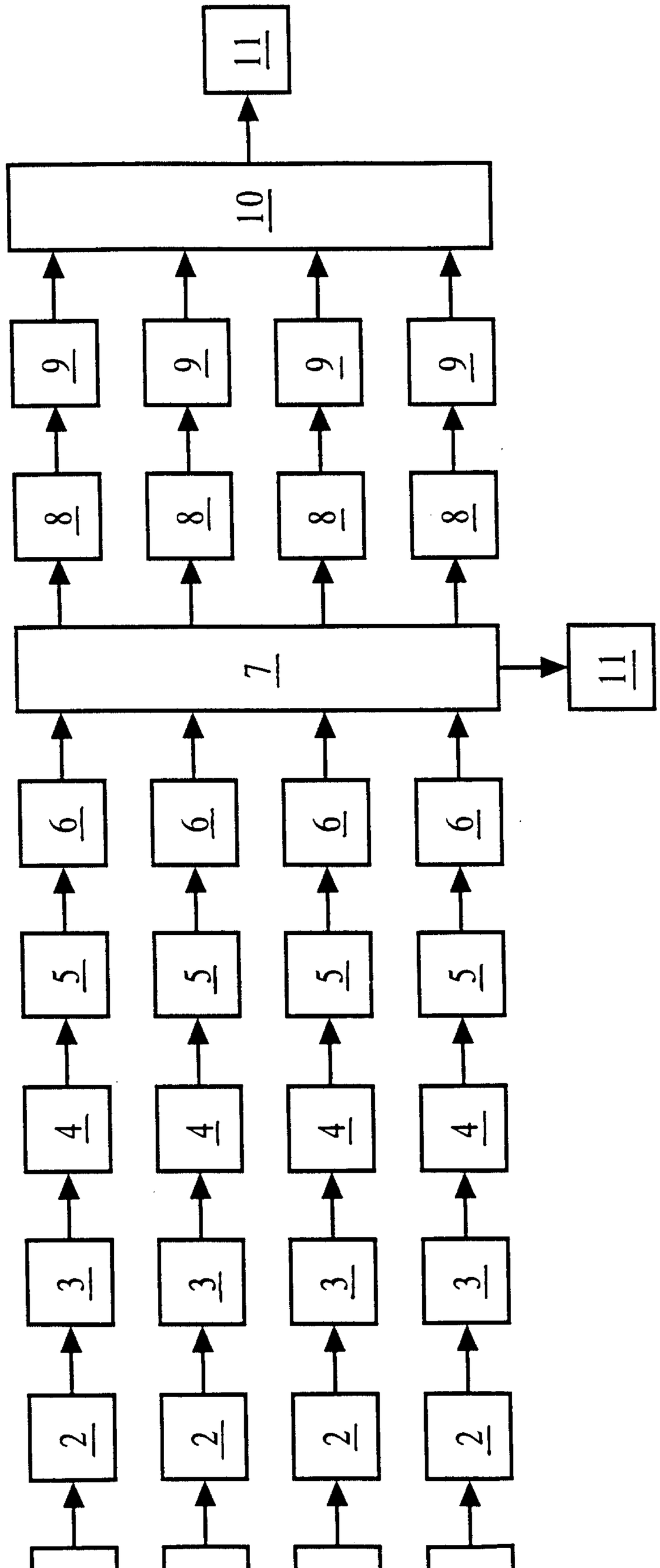


FIG. 1

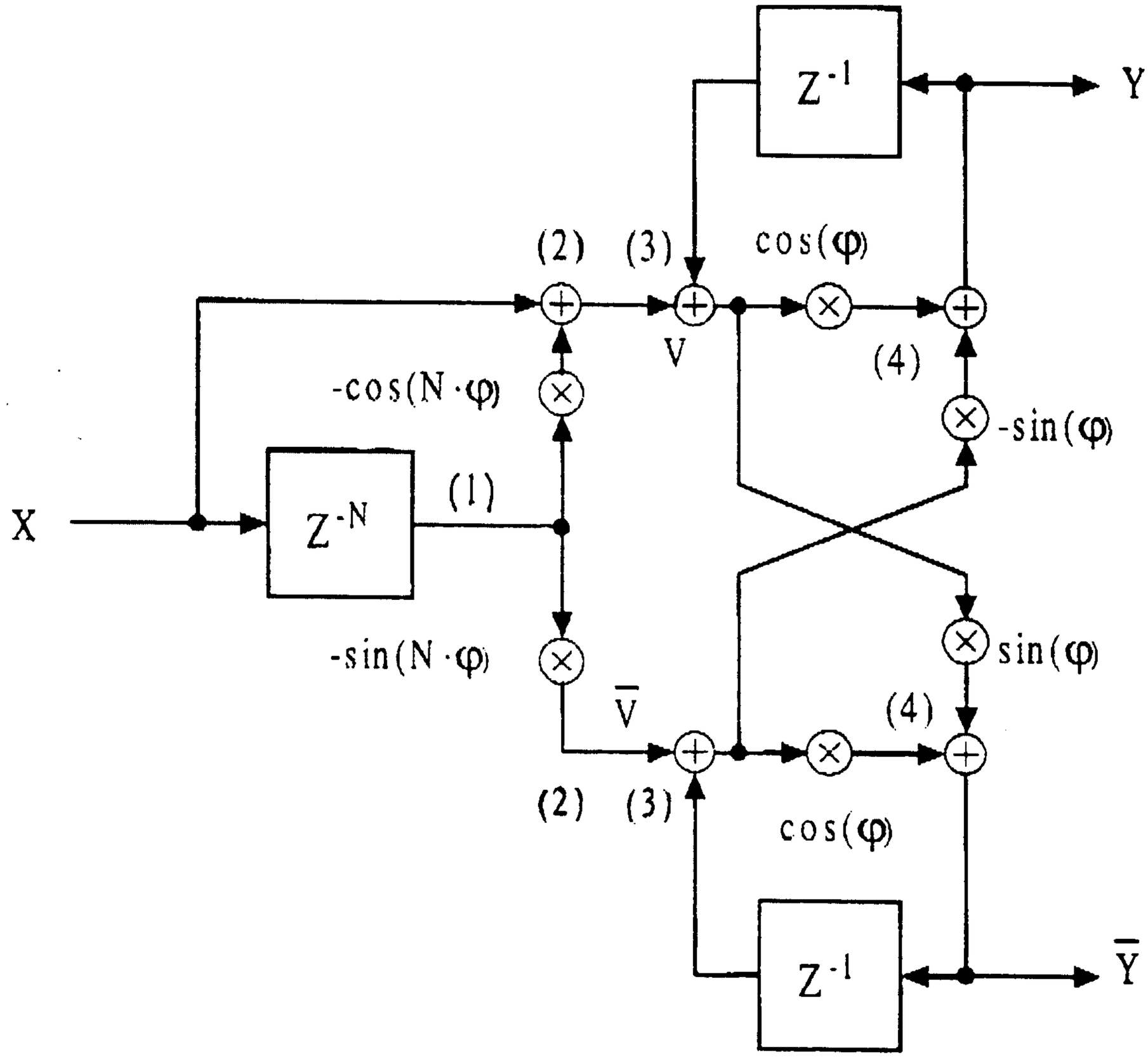

Z^{-1}
Y
(2)
(3)
cos(φ)
V
(4)
-cos(N·φ)
-sin(φ)
X
Z^{-N}
(1)
-sin(N·φ)
sin(φ)
$\overline{V}$
(4)
(2)
(3)
cos(φ)
Z^{-1}
$\overline{Y}$

FIG. 2