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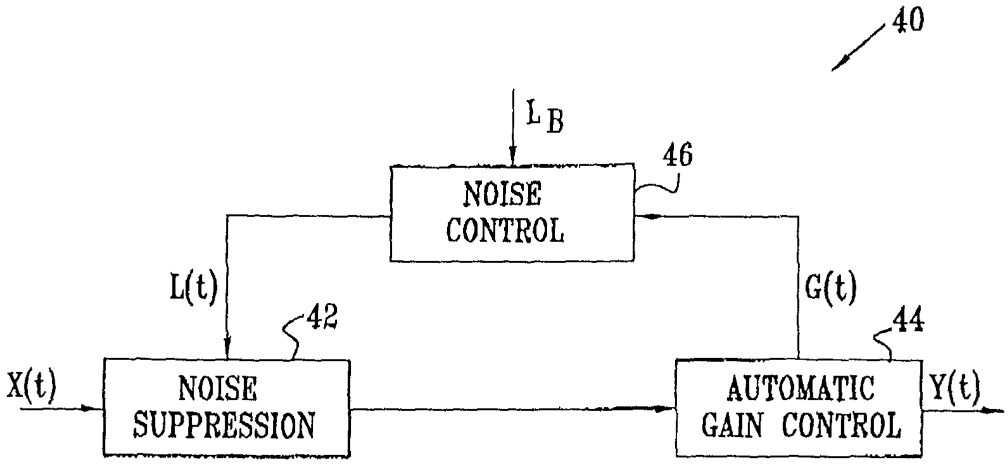
(54) Automatic gain control with noise suppression

(57) Audio processing apparatus (40) includes a a noise suppression stage (42), which applids a variable level of noise suppression to an input audio signal, so as to generate a noise-suppressed signal, and an automatic gain control (AGC) stage (44), coupled to apply a variable gain to the noise-suppressed signal, responsive to a level of the signal. A noise controller (46) receives an indication of the gain from the AGC stage and determines the level of noise suppression to be applied by the noise suppression stage responsive to the gain.

FIG. 3



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Description

CROSS-REFERENCE TO RELATED APPLICATION

[0001] This application claims the benefit of U.S. Provisional Patent Application No. 60/249,388, filed November 16, 2000, which is incorporated herein by reference.

FIELD OF THE INVENTION

[0002] The present invention relates generally to processing of audio signals, and specifically to noise reduction and automatic gain control in processing of such signals.

BACKGROUND OF THE INVENTION

[0003] Automatic gain control (AGC) is used in voice communications to compensate for differences in signal level. Such difference may arise, for example, in speakerphone applications due to the differences in distance between the microphone and several speakers participating in a teleconference. Assuming the AGC is working perfectly, the output level of the audio processing circuits should remain constant even for large variations in the input signal level received by the microphone. Unfortunately, microphones used in real environments pick up background noises. Since the level of background noise remains roughly constant, while the level of the signal varies depending on the distance of the speaker from the microphone, the signal-to-noise (s/N) level varies accordingly. When the signal is amplified or reduced by AGC to compensate for variations in the signal level, the noise level in the output signal is affected accordingly. This variable amplification of the noise level leads to an annoying effect known as "noise modulation."

[0004] Fig. 1 is a plot that schematically illustrates signals received by a microphone, representing the voices of two speakers. The first speaker (who speaks during intervals marked "A" in the figure) is about 50 cm from to the microphone, while the second speaker (speaking during intervals marked "B") is about 2 m from the microphone. Since the sound pressure level is inversely proportional to the distance from the microphone, the input level of an audio signal 20 received during the A intervals is about four times (12 dB) greater than a signal 22 during the B intervals. A background noise level 24 remains roughly constant.

[0005] Fig. 2 is a plot that schematically illustrates the result of applying AGC to the signals of Fig. 1. The AGC causes an output signal 32 during the B intervals to have a level that is roughly equal to that of an output signal 30 during the A intervals. A noise level 34 during the A intervals remains reasonably low. Strong amplification of the weak signal in the B intervals, however, causes corresponding amplification of a noise level 36 during these intervals. As a result, while the signals from both speakers are heard at approximately the same output signal level, the noise level has sharp and noticeable variations.

[0006] Digital noise suppression techniques can be used to reduce the background noise level before AGC amplification of the signal. (Noise suppression must precede AGC, since if the order of operation is reversed, variations in the AGC gain will confuse the noise suppressor's estimate of the noise level.) Common noise suppression techniques typically involve determining the noise spectrum and filtering the signal based on this spectrum in order to remove the noise components insofar as possible. Such techniques are commonly referred to as methods of "spectral attenuation" or "spectral subtraction." They are described, for example, by Boll in an article entitled "Suppression of Acoustic Noise in Speech Using Spectral Subtraction," published in *IEEE Transactions on Acoustics*, *Speech and Signal Processing,* ASSP-27, No. 2 (April, 1979), which is incorporated herein by reference.

[0007] A variety of methods of noise suppression are described in the patent literature. For example, U.S. Patent 4,185,168, to Graupe et al., whose disclosure is incorporated herein by reference, describes a system for adaptively filtering near-stationary noise from an information bearing signal. An input signal containing information as well as near-stationary noise is applied to a noise-analysis circuit and simultaneously to a noise-reduction circuit, each of which circuits comprises a plurality of bandpass filters. The background noise power is estimated by measuring an average of successive minima in each of the filters during times when substantially only noise is present. Several methods are described for determining the gain of each filter, responsive to the measured successive averaged minima and the size of the signal.

[0008] U.S. Patent 5,550,924, to Helf et al., whose disclosure is incorporated herein by reference, describes a method for reducing background noise in order to enhance speech. Properties of human audio perception are used to perform spectral and time masking to reduce perceived loudness of noise added to the speech signal. A signal is divided temporally into blocks which are then passed through a plurality of filters to remove narrow frequency band components of the noise. An estimate of the noise level in each of the filters is made by averaging measured noise powers. A FFT (Fast Fourier Transform) is performed on the blocks to determine the average noise power. Responsive to the determined noise power, a noise-reduced signal is recovered using an inverse FFT.

[0009] U.S. Patent 5,768,473, to Eatwell et al., whose disclosure is incorporated herein by reference, describes an adaptive speech filter. The filter is a modified version of that described in U.S. Patent 4,185,168, using a noise power estimate of an average of the power. The filter implements an improved adaptive spectral estimator for estimating the spectral components in a signal containing both an information signal, such as speech, and noise. Improvements over 4,185,168 relate to a noise power estimator and a computationally-efficient gain calculation method. The adaptive spectral estimator is said to be particularly suited to implementation using digital signal processing and can be used to provide improved spectral estimates of the information signal.

[0010] Generally, the amount of noise suppressed by a given noise suppressor is adjustable over a certain range. Suppression technologies known in the art can typically provide up to 8-10 dB of noise suppression with a significant improvement in sound quality. When noise suppression is increased above this level, however, noticeable distortion may be introduced in the speech signals. Therefore, in noisy environments, finding the optimal level of noise suppression involves trading off background noise against speech distortion. Referring back to the example of Figs. 1 and 2, it will be seen that if sufficient noise suppression is applied in order to eliminate the annoying noise modulation effect in the B intervals, the result will likely be undesired distortion in the audio signals in both the A and B intervals. On the other hand, if only mild noise suppression is applied as indicated by the A interval signals, noticeable noise modulation will remain,

SUMMARY OF THE INVENTION

[0011] It is an object of the present invention to provide improved methods and devices for processing of audio signals in the presence of amplitude variations and noise.

[0012] It is a further object of some aspects of the present invention to provide methods and devices for audio signal processing that reduce or eliminate noise modulation without introducing excessive distortion.

[0013] In preferred embodiments of the present invention, an audio processor comprises a noise suppression stage and an AGC stage. The amount of noise suppressed is adjusted continually according to the current AGC gain. Thus, if greater signal amplification is necessary to compensate for a drop in the signal level, more noise is suppressed compensate for residual noise amplification in the output signal from the audio processor. On the other hand, when the signal level increases, the noise suppression is reduced in order to eliminate possible distortion. The audio processor can thus be adjusted to give optimal audio quality, balancing noise modulation against signal distortion, over a range of different signal levels.

[0014] There is therefore provided, in accordance with a preferred embodiment of the present invention, audio processing apparatus, including:

> a noise suppression stage, adapted to apply a variable level of noise suppression to an input audio signal, so as to generate a noise-suppressed signal;
> an automatic gain control (AGC) stage, coupled to determine a variable gain responsive to a level of the noise-suppressed signal, and to apply the gain to the noise-suppressed signal so as to generate an amplified output signal; and
> a noise controller, coupled to receive an indication of the gain from the AGC stage and to determine the level of noise suppression to be applied by the noise suppression stage responsive to the gain.

[0015] Preferably, the noise suppression stage is adapted to apply spectral compression to the input audio signal.

[0016] Additionally or alternatively, the noise controller is adapted to determine the level of noise suppression as a monotonically-increasing function of the gain. Preferably, the level of noise suppression determined by the noise controller increases in proportion to a power of the gain, wherein the power is less than or equal to one. Most preferably, the level of noise suppression $L(t)$ is given substantially by an expression of the form $L(t) = L_B + (G(t)^x$, wherein $G(t)$ is the gain, L_B is an additive factor, and x is a number less than or equal to one.

[0017] In a preferred embodiment, the AGC stage is adapted to increase and decrease the gain in alternation in response to alternations in the level of the noise-suppressed signal due to receiving the input audio signal from alternating weak and strong audio sources, respectively, and the noise Controller is adapted to decrease and increase the level of noise suppression, responsive respectively to the gain increasing and decreasing.

[0018] There is also provided, in accordance with a preferred embodiment of the present invention, a method for audio processing, including:

> suppressing noise in applying an input audio signal using a variable level of noise suppression, so as to generate a noise-suppressed signal;
> determining a variable gain responsive to a level of the noise-suppressed signal;
> applying the gain to the noise-suppressed signal so as to generate an amplified output signal; and
> determining the level of noise suppression to be applied to the input audio signal responsive to the gain.

[0019] The present invention will be more fully understood from the following detailed description of the preferred embodiments thereof, taken together with the drawings in which:

BRIEF DESCRIPTION OF THE DRAWINGS

[0020]

> Fig. 1 is a plot that schematically illustrates signals received by a microphone;

Fig. 2 is a plot that schematically illustrates the signals of Fig. 1 following AGC amplification, as is known in the art;

Fig. 3 is a block diagram that schematically illustrates an audio processor, in accordance with a preferred embodiment of the present invention; and

Fig. 4 is a plot that schematically illustrates signals output by the audio processor of Fig. 3, in accordance with a preferred embodiment of the present invention.

DETAILED DESCRIPTION OF PREFERRED EMBODIMENTS

[0021] Fig. 3 is a block diagram that schematically illustrates an audio processor 40, in accordance with a preferred embodiment of the present invention. The audio processor receives a raw audio input signal $X(t)$, from a microphone, for example (not shown), and outputs a processed audio signal $Y(t)$. Audio processor comprises a noise suppression stage 42, followed by an AGC stage 44. Preferably, $X(t)$ and $Y(t)$ are analog signals, and stages 42 and 44 are implemented using suitable analog circuit elements, such as tunable filters and variable-gain amplifiers, as are known in the art. Alternatively, $X(t)$ and $Y(t)$ are digitized, and the processing functions described hereinbelow are implemented using digital logic circuits. Mixed analog and digital implementations may also be used.

[0022] Preferably, noise suppression stage 42 implements a method for suppressing near-stationary noises and tones described in U.S. Patent Application 09/605,174, filed June 28, 2000, which is assigned to the assignee of the present patent application and whose disclosure is incorporated herein by reference. According to this method, the input noise is divided into multiple frequency bands, and the maximum and minimum noise levels in each band are determined over a period of time. Based on these noise levels, a gain is computed in each band using spectral subtraction and/or spectral compression. Preferably, given an effective level of the signal $\hat{A}^a_n$, and a difference between the upper and lower noise estimates Δ^a_n, the noise suppression gain G_{NS} is given by the following formulas:

$$\text{If } \hat{A}^a_n \le \Delta^a_n, \text{ then } G_{NS} = G_{\min}. \qquad (1)$$

$$\text{If } \hat{A}^a_n > \Delta^a_n \text{ and } \hat{A}^a_n < (S+1)\Delta^a_n,$$

$$\text{then } G_{NS} = G_{\min} + (1 - G_{\min})\left(\frac{\hat{A}^a_n - \Delta^a_n}{S \cdot \Delta^a_n}\right)^2. \qquad (2)$$

$$\text{If } \hat{A}^a_n \ge (S+1) \cdot \Delta^a_n, \text{ then } G_{NS} = 1, \qquad (3)$$

wherein G_{min} is a minimum value of the gain G_{NS}.

[0023] The gains G_{NS} are applied by noise suppression stage 42 to the respective frequency bands of the input signal $X(t)$ to generate a noise-suppressed input to AGC stage 44. Alternatively, the noise suppression stage may employ other techniques, such as those described in the Background of the Invention, or substantially any other suitable noise suppression method known in the art.

[0024] As can be seen in Fig. 3, AGC stage 44 operates on the audio signals after processing by noise suppression stage 42. The AGC stage determines a variable gain $G_{AGC}(t)$ to be applied to the signals in order to compensate for variations in the input signal level, The current value of $G_{AGC}(t)$ is provided to a noise control block 46. Based on this value, the noise control block computes the amount of noise suppression $L(t)$ to be applied by noise suppression stage 42 to the input signal $X(t)$. Preferably, the values of parameters used in noise suppression stage 42, such as G_{min}, are continually adjusted so that the total amount of noise suppression is equal to the current value of $L(t)$. Although block 46 is shown in the figure as a separate entity for the sake of clarity of explanation, those skilled in the art will appreciate that the function of this block may alternatively be integrated into either stages 42 or stage 44.

[0025] Preferably, $L(t)$ is determined based on the current AGC gain $G_{AGC}(t)$ and on a basic noise suppression level L_B, which corresponds to the amount of noise suppressed in the output signal when AGC gain is equal to unity. Various functions may be used to relate $L(t)$ to $G_{AGC}(t)$ and L_B. Preferably, $L(t)$ increases monotonically relative to both $G_{AGC}(t)$ and L_B . For example, the following function provides noise suppression with full compensation for changes in the AGC gain:

$$L(t) = L_B + G_{AGC}(t) \qquad (4)$$

(The noise suppression levels and AGC gain are specified here in decibels.) It is seen that using equation (4), the noise is first suppressed by $L(t)$ decibels and than expanded by $G_{AGC}(t)$ decibels. Thus, the noise is always suppressed by the original amount of L_B decibels.

[0026] The function of equation (4) may not be optimal, however, when large variations in the input signal level can occur, as it may lead to excessive noise suppression, with noticeable distortions in the output signal $Y(t)$. For example, if the basic noise suppression level L_B is 5 dB and AGC gain is 15 dB, then the total amount of noise suppression will be 20 dB. Under such conditions, a milder dependence between the AGC gain $G_{AGC}(t)$ and noise suppression level $L(t)$ is preferable, such as a dependence of $L(t)$ on a fractional power of the gain $(G_{AGC}(t))^x$, with $x < 1$. For example, the following function provides a good compromise between modulation of the noise level in the output signal and the output speech quality:

$$L(t) = L_B + \sqrt{G_{AGC}(t)} \qquad (5)$$

Greater or smaller fractional powers of $G_{AGC}(t)$ may also be used. Alternative functions will be apparent to those skilled in the art.

[0027] Fig. 4 is a plot that schematically shows the output signal $Y(t)$ obtained by operating on the input signal $X(t)$ shown in Fig. 1 using audio processor 40, in accordance with a preferred embodiment of the present invention. The variable noise suppression $L(t)$ is given by equation (5). Signals 50 and 52 during intervals A and B, respectively, are amplified by AGC stage 44 to give comparable levels of perceptual loudness. Respective noise levels 54 and 56 are suppressed during both intervals A and B, as well. The level of noise suppression during the B intervals is greater than that during the A intervals, but due to the square root factor in equation (5), there is still slightly more residual noise in the B intervals. The parameters governing the dependence of $L(t)$ on $G(t)$ are preferably chosen and adjusted based on the background noise and signal conditions so as to balance the residual noise modulation against distortion effects due to the noise suppression, in a way that gives the most pleasing perceived sound quality.

[0028] It will be appreciated that the preferred embodiments described above are cited by way of example, and that the present invention is not limited to what has been particularly shown and described hereinabove. Rather, the scope of the present invention includes both combinations and subcombinations of the various features described hereinabove, as well as variations and modifications thereof which would occur to persons skilled in the art upon reading the foregoing description and which are not disclosed in the prior art.

Claims

1. Audio processing apparatus, comprising:

a noise suppression stage, adapted to apply a variable level of noise suppression to an input audio signal, so as to generate a noise-suppressed signal;
an automatic gain control (AGC) stage, coupled to determine a variable gain responsive to a level of the noise-suppressed signal, and to apply the gain to the noise-suppressed signal so as to generate an amplified output signal; and
a noise controller, coupled to receive an indication of the gain from the AGC stage and to determine the level of noise suppression to be applied by the noise suppression stage responsive to the gain.

2. Apparatus according to claim 1, wherein the noise controller is adapted to determine the level of noise suppression as a monotonically-increasing function of the gain.

3. Apparatus according to claim 2, wherein the level of noise suppression determined by the noise controller increases in proportion to a power of the gain, wherein the power is less than or equal to one.

4. Apparatus according to claim 3, wherein the level of noise suppression $L(t)$ is given substantially by an expression of the form $L(t) = L_B + (G(t))^x$, wherein $G(t)$ is the gain, L_B is an additive factor, and x is a number less than or equal to one.

5. Apparatus according to claim 1, wherein the AGC stage is adapted to increase and decrease the gain in alternation in response to alternations in the level of the noise-suppressed signal due to receiving the input audio signal from alternating weak and strong audio sources, respectively, and wherein the noise controller is adapted to decrease and increase the level of noise suppression, responsive respectively to the gain increasing and decreasing.

6. A method for audio processing, comprising:

suppressing noise in applying an input audio signal using a variable level of noise suppression, so as to generate a noise-suppressed signal;
determining a variable gain responsive to a level of the noise-suppressed signal;
applying the gain to the noise-suppressed signal so as to generate an amplified output signal; and
determining the level of noise suppression to be applied to the input audio signal responsive to the gain.

7. A method according to claim 6, wherein the level of noise suppression is determined as a monotonically-increasing function of the gain.

8. A method according to claim 7, wherein the level of noise suppression is proportional to a power of the gain, wherein the power is less than or equal to one.

9. A method according to claim 8, wherein the level of noise suppression $L(t)$ is given substantially by an expression of the form $L(t) = L_B + (G(t))^x$, wherein $G(t)$ is the gain, L_B is an additive factor, and x is a number less than or equal to one.

10. A method according to claim 6, wherein determining the variable gain comprises increasing and decreasing the gain in alternation in response to alternations in the level of the noise-suppressed signal due to receiving the input audio signal from alternating weak and strong audio sources, and wherein determining the level of noise suppression comprises decreasing and increasing the level of noise suppression responsive to the increasing and decreasing of the gain.

FIG. 1
PRIOR ART

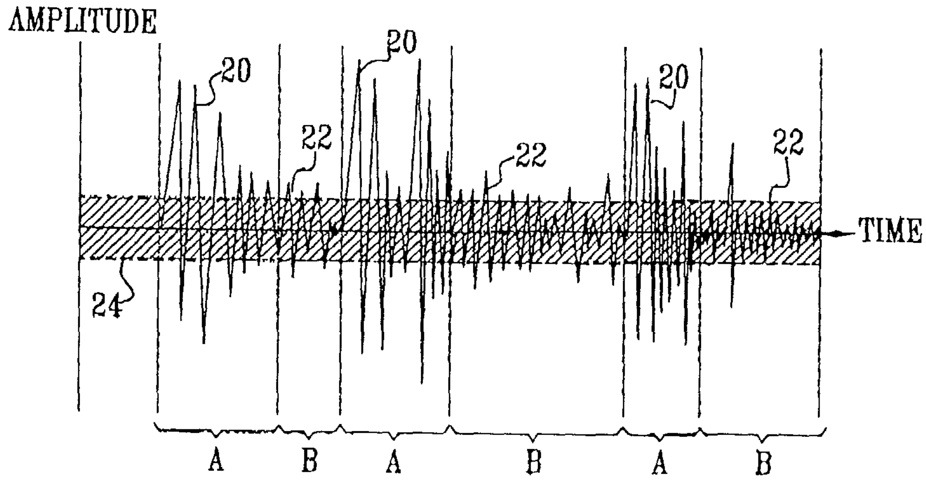


FIG. 2
PRIOR ART

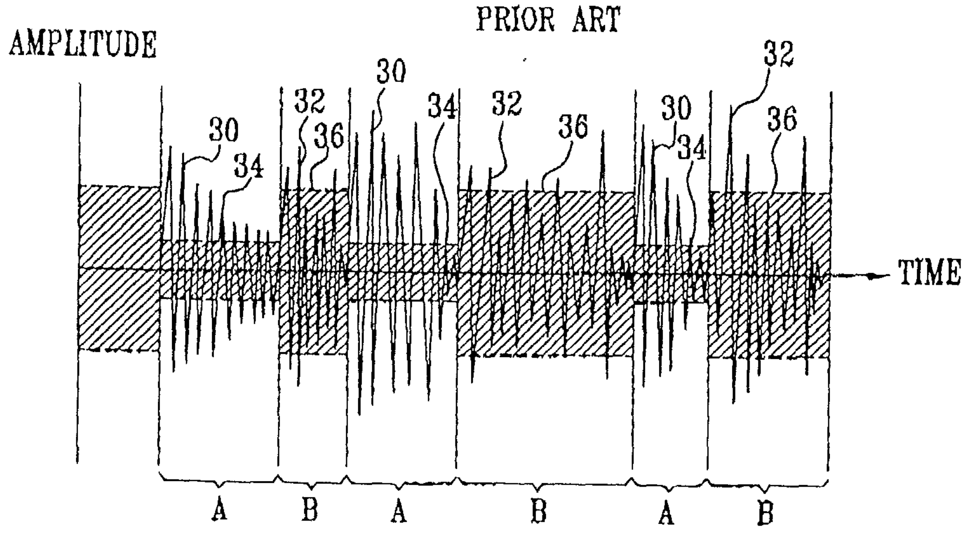


FIG. 3

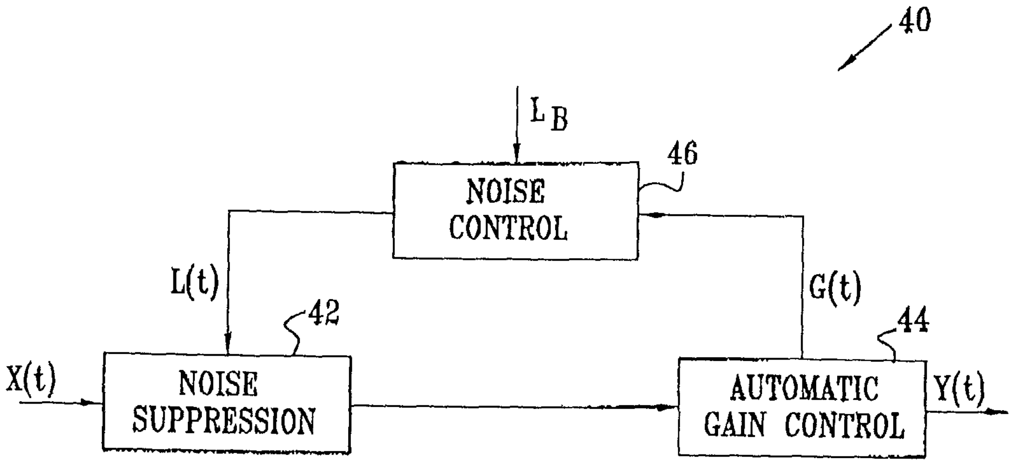
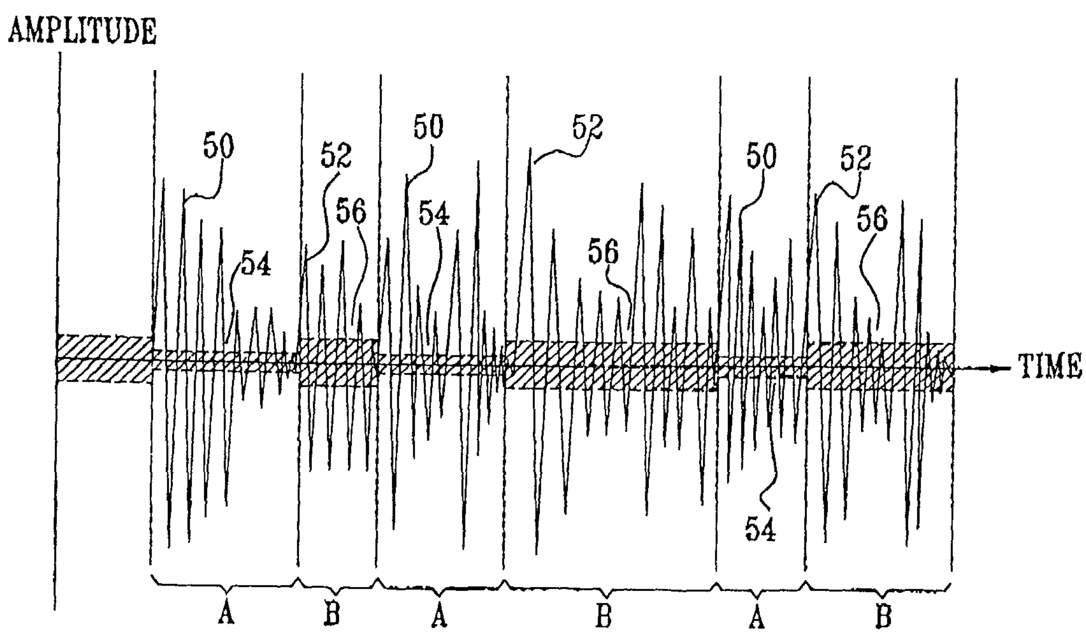

40
L_B
NOISE CONTROL
46
L(t)
G(t)
42
44
X(t)
NOISE SUPPRESSION
AUTOMATIC GAIN CONTROL
Y(t)

FIG. 4


AMPLITUDE
50
52
56
54
TIME
A
B
A
B
A
B