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(54) **Method for processing an acoustic input signal to provide an output signal with reduced noise**

(57) The invention provides a method and an apparatus for processing an acoustic input signal to provide an output signal with reduced noise, comprising weight-

ing the input signal using a frequency dependent weighting function, wherein the weighting function is bounded below by a frequency dependent threshold function.

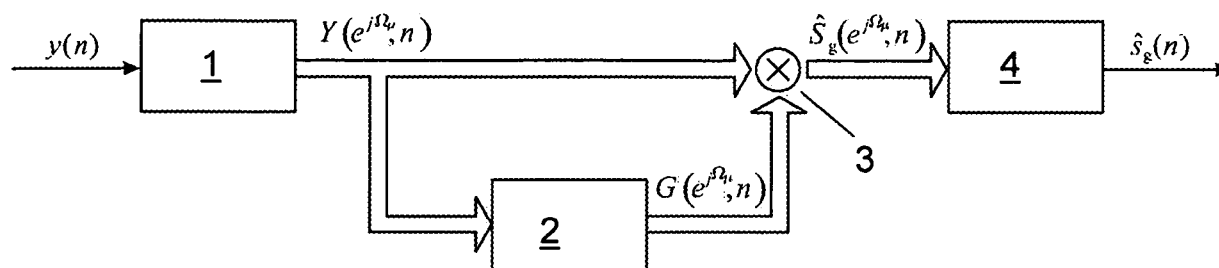


Fig. 1

Description

[0001] The present invention is concerned with a method and an apparatus for processing an acoustic input signal to provide an output signal with reduced noise.

5 **[0002]** Noise suppression in acoustic signals is an important issue in different fields. For example, handsfree telephony systems in many cases rely on different noise suppression methods which is particularly useful if such a handsfree system is used in a noisy environment such as in a vehicular cabin. In such a case, the wanted signal, namely the speech signal, is disturbed by various interferences stemming from different noise sources such as loudspeakers or noise produced by the moving vehicle.

10 **[0003]** Furthermore, also in the case of speech recognition systems, in which speech commands are used to control specific devices which might also be implemented in a vehicular environment, noise suppression provides useful in order to reduce mis-recognitions.

15 **[0004]** Common methods for noise suppression involve, for example, so-called Wiener filter (see E. Hänsler, G. Schmidt, "Acoustic Echo and Noise Control", Wiley, Hoboken, NJ, USA, 2004) or spectral subtraction (see P. Vary, R. Martin, "Digital Speech Transmission", Wiley, Hoboken, NJ, USA, 2006). Other prior art methods are known, for example, from K. Linhard, T. Haulick, "Spectral Noise Subtraction with Recursive Gain Curves", ICSLP '98, Conference Proceedings, Number 4, Pages 1479 - 1482 or H. Puder, O. Soffke, "An Approach for an Optimized Voice-Activity Detector for Noisy Speech Signals", EUSIPCO '02, Conference Proceedings Number 1, Pages 243 - 246.

20 **[0005]** Known noise suppression methods suffer from the drawback that the noise suppression is rather inflexible, for example, in that changing environmental conditions are hardly taken into account. In view of this, it is a problem underlying the invention to provide a method and an apparatus for processing an acoustic input signal to provide an output signal with reduced noise showing more flexibility. This problem is solved by a method according to claim 1 and an apparatus according to claim 19.

25 **[0006]** Accordingly, the invention provides a method for processing an acoustic input signal to provide an output signal with reduced noise, comprising weighting the input signal using a frequency dependent weighting function, wherein the weighting function is bounded below by a frequency dependent threshold function.

30 **[0007]** Due to this frequency dependent weighting function with a frequency dependent threshold function as lower bound, it surprisingly turned out that a much more flexible noise suppression can be achieved. In principle, the input signal may stem from an arbitrary source, such as a microphone or a microphone array of a handsfree system. The input signal particularly comprises a wanted signal component and a noise signal component, the latter representing a disturbance in the signal. The input signal may be provided in digital form.

35 **[0008]** Weighting the input signal with the weighting function may be achieved by multiplying the input signal with the weighting function. In principle, the input signal may have passed one or more filter stages (for example, a beamformer and/or a bandpass filter) before performing the weighting. After the weighting, one or more filters may be provided before the final output signal with reduced noise is obtained.

[0009] In the above method, the threshold function may be a time dependent function. In this way, an adaptation not only to different frequencies but also to time varying conditions may be achieved.

40 **[0010]** The above methods may comprise adapting the weighting function. In particular, the methods may comprise performing wanted signal detection and adapting the weighting function if no wanted signal is detected. In this way, the adaptation to changing conditions is obtained, thus, further improving noise suppression.

[0011] Adapting the weighting function may comprise adapting the power of the weighting function; in particular, adapting the weighting function may be limited to adapting the overall power of the weighting function. Thus, except for the overall power (i.e. the power over the whole frequency range), the weighting function is not modified. The adapting may be performed with respect to the overall power of the input signal.

45 **[0012]** Wanted signal detection may be performed in different ways. For example, common voice activity detectors may be used. In principle, adapting the weighting function may also be performed without such wanted signal detection; in such a case, for example, minimum statistics may be used.

50 **[0013]** The threshold function may be based on a target noise spectrum. In this way, the residual noise, i.e., the noise in the output signal after the weighting step, may be controlled in a desired way. Thus, the method may be configured such that the residual noise approaches or converges to the target noise spectrum according to a predetermined criterion or measure.

55 **[0014]** The target noise spectrum may be time dependent. In this way, the target noise spectrum may be adapted to varying conditions, particularly regarding any background noise. A time dependent target noise spectrum may be obtained by providing a time independent initial target noise spectrum and adapting or modifying the initial target noise spectrum according to a predetermined criterion. Such an adaptation may be performed, for example, using a predetermined adaptation factor which may be time dependent.

[0015] The method may comprise adapting the target noise spectrum. In particular, it may comprise performing wanted signal detection and adapting the target noise spectrum if no wanted signal is detected. Adapting the target noise

spectrum may comprise adapting the overall power of the target noise spectrum; in particular, adapting the target noise spectrum may be limited to adapting the overall power of the target noise spectrum. The adapting may be performed with respect to the overall power of the input signal.

5 [0016] In particular, the target noise spectrum at time n may be incremented if the power of the target noise spectrum at time $(n-1)$ within a predetermined frequency interval is smaller than a predetermined attenuation factor times the power of an estimate of a noise component in the input signal at time n within the predetermined frequency interval.

[0017] Incrementing the target noise spectrum may comprise multiplying the target noise spectrum with a predetermined incrementing factor; this incrementing factor will be greater than one. An estimate of the power of a noise component in the input signal may be obtained by temporally smoothing the current subband power of the input signal; alternatively, 10 minimum statistics may be used. Here, n denotes the discrete time variable.

[0018] The target noise spectrum and time n may be decremented if the power of the target noise spectrum at time $(n-1)$ within a predetermined frequency interval is greater than or equal to a predetermined attenuation factor times an estimate of the power of a noise component in the input signal at time n within the predetermined frequency interval. Decrementing the target noise spectrum may be performed by multiplying the target noise spectrum with a predetermined 15 decrementing factor. The predetermined attenuation factor and/or the predetermined frequency interval for the decrementing step may be equal to the respective attenuation factor and frequency interval for the incrementing step.

[0019] In this way, an adaptation to the overall power of the input signal is obtained; however, the general form of the target noise spectrum is not changed.

20 [0020] Wanted signal detection may be performed, for example, by comparing the weighting function averaged over a predetermined frequency interval at time $(n-1)$ and a predetermined threshold value. Particularly if the threshold value is exceeded, an adaptation may take place.

[0021] The threshold function may be based on the minimum of a predetermined minimum attenuation value and a quotient of the target noise spectrum and the absolute value of the input signal. This allows taking into account the current power of the input signal, and providing a suitable minimal weighting, thus, a suitable attenuation or damping. 25 In particular, the threshold function may be equal to this minimum.

[0022] The threshold function may be based on the maximum of said minimum and a predetermined maximum attenuation value. Thus, suitable upper and lower bounds (being time dependent) are obtained. In particular, the threshold function may be equal to this maximum.

30 [0023] The threshold function at time n may be based on a convex combination of the threshold function at time $(n-1)$ and said maximum at time n . This results in a more natural residual noise. A convex combination is a linear combination in which the coefficients are non-negative and some up to one. Thus, the threshold function obtained in this way is more based on a recursive smoothing. In particular, the threshold function at time n may be equal to this convex combination.

[0024] In the above-described methods, the threshold function may be based on at least two target noise spectra. The use of more than one target noise spectrum allows to distinguish between different ambient conditions and to adapt 35 the method accordingly. For example, in the case of noise suppression for a handsfree system in a vehicular cabin, a first noise spectrum may be used for lower speed of the vehicle (i.e., below a predetermined threshold), and a second target noise spectrum may be used for higher speed.

[0025] The weighting function may be based on the maximum of the threshold function and a predetermined filter characteristic. In this way, an advantageous weighting function with a lower bound is obtained. In particular, the filter characteristic alone need not be restricted to values above a certain threshold. 40

[0026] The filter characteristic may be time dependent. Thus, an adaptation to the ambient condition is possible.

[0027] In the above-described methods, the weighting function may be based on a Wiener characteristic. In particular, the above-mentioned filter characteristic may be a Wiener characteristic. Alternatively, the weighting function may be based on other filter characteristics, for example, based on the Ephraim-Malah algorithm or the Lotter algorithm.

45 [0028] The above-described methods may be performed in the frequency domain. In particular, at least one of the steps may be performed for each frequency subband separately. For example, adapting the target noise spectrum and/or determining the above-mentioned minima and/or maxima may be performed for each frequency subband.

[0029] In particular, the method may comprise passing an input signal through an analysis filter bank. For example, a DFT (Discrete Fourier Transform) or DCT (Discrete Cosine Transform), a polyphase filter bank or a gammatone filter bank may be used. With such an analysis filter bank, a separation into frequency subbands or short-time spectra may be obtained. 50

[0030] In the previously described methods, the weighting function may be based on an estimated power density spectrum of a noise signal component and/or an estimated power density spectrum of the input signal. In particular, the weighting function may be based on a quotient of these power density spectra.

55 [0031] The estimated power density spectrum of a noise signal component may be determined as indicated above. The estimated power density spectrum of the input signal may be determined as the absolute value squared of a vector containing the current subband input signals as coefficients.

[0032] The invention also provides a computer program product comprising one or more computer readable media

having computer-executable instructions for performing the steps of the above described methods when run on a computer.

[0033] Furthermore, the invention provides an apparatus for processing an acoustic input signal to provide an output signal with reduced noise, comprising means for weighting the input signal using a frequency dependent weighting function, wherein the weighting function is bounded below by a frequency dependent threshold function.

[0034] In particular, the apparatus may comprise means for performing the steps of the above described methods. For example, the apparatus may comprise means for adapting the weighting function.

[0035] Further aspects of the invention will be described in the following with reference to the Figures and illustrative embodiments.

Figure 1 schematically illustrates an example of the structure of a system for providing an output signal with reduced noise;

Figure 2 examples of time-frequency analyses for an output signal with reduced noise according to the invention; and

Figure 3 illustrates further examples of time-frequency analyses relating to the method to provide an output signal with reduced noise.

[0036] Figure 1 illustrates schematically an example of the structure of a system to perform a noise reduction method. Such a system may be implemented, for example, in handsfree telephony systems or handsfree speech recognition systems which may be used in a vehicular cabin. Typically, an acoustic signal is recorded by one or more microphones resulting in a discretized microphone signal $y(n)$. It is to be understood that the signal $y(n)$ may have passed one or more filters before arriving at the noise suppression stage as illustrated. Here and in the following, n denotes the time index.

[0037] The input signal $y(n)$ is composed of a wanted signal component $s(n)$ and a noise component $b(n)$;

$$y(n) = s(n) + b(n).$$

[0038] In many cases, the wanted signal component is a speech signal. In the example shown, the processing of the input signal is performed in the frequency domain. For this purpose, an analysis filter bank 1 is provided so that input subband signals or short-time spectra $Y(e^{j\Omega_\mu}, n)$ are obtained. Ω_μ are the discrete frequency sampling points as determined by the analysis filter bank, wherein

$$\mu \in \{0, 1, \dots, M - 1\}.$$

[0039] The analysis filter bank 1 may be based on a DFT (Discrete Fourier Transform) or a DCT (Discrete Cosine Transform); or alternatively, polyfaced filter banks or gammatone filter banks (see P.P. Vaidyanathan, "Multirate Systems and Filter Banks", Prentice Hall, Englewood Cliffs, NJ, USA, 1992) may be used. Every r cycles, the subband signals are determined anew.

[0040] As an example, the number of subbands M may be 256 and the frame displacement r may be 64. As window function, a Hann window having a length of 256 may be employed. However, it is to be noted that other filter bank parameters may be used as well.

[0041] In block 2, for each subband, a weighting function (sometimes also called attenuation factors or damping factors) $G(e^{j\Omega_\mu}, n)$ are to be determined. This weighting function is both time (n) and frequency (Ω_μ) dependent. The weighting function is then used to weight the input subband signals $Y(e^{j\Omega_\mu}, n)$ in block 3 via a multiplication

$$\hat{S}_g(e^{j\Omega_\mu}, n) = Y(e^{j\Omega_\mu}, n)G(e^{j\Omega_\mu}, n).$$

[0042] The subband signals $\hat{S}_g(e^{j\Omega_\mu}, n)$ are estimates for the undisturbed wanted subband signals $S(e^{j\Omega_\mu}, n)$. These estimates are then combined in a synthesis filter bank 4 to obtain an output signal $\hat{S}_g(n)$.

[0043] According to the present invention, an initial power density spectrum of a target noise $S_{bb,target}(e^{j\Omega_\mu})$ is provided. This initial power density spectrum may be a melodic noise as obtained via comparison tests, for example. Alternatively, it may correspond to the noise which had been used to train a speech recognition system. In this case, the speech recognition system will be used both in the training phase and a operation phase with the same residual noise.

[0044] Based on this initial target noise power density spectrum, a real value target noise vector for the starting time ($n = 0$) is determined:

$$B_{target}(e^{j\Omega_\mu}, 0) = \sqrt{S_{bb,target}(e^{j\Omega_\mu})}$$

[0045] The overall amplification or power of the target noise will be adapted to the current background noise conditions. For this, speech activity detection is performed. This may take place using common speech activity detectors. A multiplicative adaptation is performed for those signal frames for which in the preceding frame no speech activity had been detected. However, if speech activity had been detected, no adaptation of the target noise will take place:

$$B_{target}(e^{j\Omega_\mu}, n) = \begin{cases} \Delta_B(n) B_{target}(e^{j\Omega_\mu}, n-1), & \frac{1}{M} \sum_{\mu=0}^{M-1} G(e^{j\Omega_\mu}, n-1) > K_G, \\ B_{target}(e^{j\Omega_\mu}, n-1), & \text{else.} \end{cases}$$

[0046] In this example, thus, speech activity is detected by comparing the mean attenuation factor (weighting function)

$$\frac{1}{M} \sum_{\mu=0}^{M-1} G(e^{j\Omega_\mu}, n-1) \text{ (i.e., averaged over all frequency samples) of the last signal frame with a predetermined}$$

threshold value K_G . The determination of the weighting function G (used to determine the mean attenuation factor) will be described in detail below. For the constant K_G , a value of 0.5 may be used.

[0047] The correction factor $\Delta_B(n)$ is determined as follows. Firstly, an attenuation value K_B is provided corresponding to the amount the target noise has fallen below the current noise within a predefined frequency interval. As an example, the frequency interval may have a lower bound of

$$\Omega_{\mu_0} = 400 \text{ Hz}$$

and an upper bound of

$$\Omega_{\mu_1} = 700 \text{ Hz} .$$

[0048] The attenuation value may be

$$K_B = 0.13 .$$

[0049] This corresponds to an attenuation of 18 dB. Having determined these parameters, the multiplicative correction may be determined using

$$\Delta_B(n) = \begin{cases} \Delta_{\text{ink}}, & \text{if } \sum_{\mu=\mu_0}^{\mu_1} B_{\text{target}}^2(e^{j\Omega_\mu}, n-1) < K_B \sum_{\mu=\mu_0}^{\mu_1} \hat{S}_{bb}(\Omega_\mu, n), \\ \Delta_{\text{dec}}, & \text{else,} \end{cases}$$

[0050] the incrementing constant Δ_{ink} and the decrementing constant Δ_{dec} fulfill:

$$0 \ll \Delta_{\text{dec}} < 1 < \Delta_{\text{ink}} \ll \infty.$$

[0051] As an example,

$$\Delta_{\text{dec}} = 0.98$$

$$\Delta_{\text{ink}} = 1.02$$

[0052] In this way, the form of the target noise (over the frequency range) will not be changed. However, the overall power is adapted. This adaptation is quite slowly so that short or fast variations of the estimated power density spectrum $\hat{S}_{bb}(\Omega_\mu, n)$ are not transferred to the target noise.

[0053] In the above equation, the estimated power density spectrum of the noise $\hat{S}_{bb}(\Omega_\mu, n)$ may be determined using a temporal smoothing of the subband powers of the current input signal. Such a smoothing is performed only during speech pauses whereas during speech activity, no smoothing will take place. Alternatively, a minimum statistics may be performed for which no speech pause detection is required (see, for example, R. Martin, "Noise Power Spectral Density Estimation Based on Optimal Smoothing and Minimum Statistics", IEEE Trans. Speech Audio Process., Volume T-SA-9, Number 5, Pages 504 - 512, 2001).

[0054] With this power adjusted target noise, an interim maximal attenuation may be determined as

$$\tilde{G}_{\text{min}}(e^{j\Omega_\mu}, n) = \max \left\{ G_1, \min \left\{ G_0, \frac{B_{\text{target}}(e^{j\Omega_\mu}, n)}{|Y(e^{j\Omega_\mu}, n)|} \right\} \right\}$$

due to the determination of the minimum and the constant G_0 (minimum attenuation value) it is guaranteed that at least an attenuation of G_0 is always present. Determining the maximum and using the constant G_1 (maximum attenuation value), the maximal attenuation is bounded. As an example, the minimum attenuation value is chosen to be

$$G_0 = 0.5.$$

[0055] In other words, the minimal attenuation is about 6 dB. The maximum attenuation value may be chosen to be

$$G_1 = 0.05$$

5 corresponding to an attenuation of about 26 dB.

[0056] If this interim maximal attenuation (corresponding to a lower bound for the weighting function) were used for a noise reduction characteristic, a tonal residual noise would occur. This is because only small variations in the absolute value of the output signal are allowed and only the phase is varied. This may result in an unnatural sound.

10 **[0057]** This may be avoided using artificial level variations, for example, via a random number generator. Another possibility is to use the temporary level variations of the disturbed input signal (at least partly). This may be done via a recursive smoothing of the interim maximal attenuation:

$$15 \quad G_{\min}(e^{j\Omega\mu}, n) = \gamma G_{\min}(e^{j\Omega\mu}, n-1) + (1-\gamma)\tilde{G}_{\min}(e^{j\Omega\mu}, n).$$

20 **[0058]** For the constant γ used for the coefficients in this convex combination, one has:

$$0 \leq \gamma \leq 1.$$

25 **[0059]** If γ is very small, only some level variations will occur. In this case, the residual noise will be tonal but will largely correspond to the target noise spectrum. In the case of a large γ , a more natural residual noise will be obtained, however, a correspondence with the target noise is only given for medium and large time intervals. As an example, one may choose

$$30 \quad \gamma = 0.7.$$

35 **[0060]** In this way, an adaptive attenuation bound or lower threshold function is obtained which may be used in different kinds of characteristics for noise suppression. In case of a Wiener characteristic, the weighting of the input signal may be performed using:

$$40 \quad G(e^{j\Omega\mu}, n) = \max \left\{ G_{\min}(e^{j\Omega\mu}, n), 1 - \beta(e^{j\Omega\mu}, n) \frac{\hat{S}_{bb}(\Omega_\mu, n)}{\hat{S}_{yy}(\Omega_\mu, n)} \right\}.$$

45 **[0061]** In this equation, $\hat{S}_{yy}(\Omega_\mu, n)$ denotes the estimated power density spectrum of the input signal. For this estimate, one may use:

$$50 \quad \hat{S}_{yy}(\Omega_\mu, n) = \left| Y(e^{j\Omega\mu}, n) \right|^2.$$

55 **[0062]** The noise overestimation factor $\beta(e^{j\Omega\mu}, n)$ may be time and frequency dependent, for example, as disclosed in the article by K. Linhard, T. Haulick.

[0063] It is to be noted that the threshold function determined in this way need not be used in the context of a Wiener characteristic. In particular, other characteristics such as in the Ephraim-Malah algorithm (see Y. Ephraim, D. Malah,

"Speech Enhancement Using a Minimum Mean-Square Error Short-Time Spectral Amplitude Estimator", IEEE Trans. Acoust. Speech Signal Process., Volume 32, Number 6, Pages 1109 - 1121, 1984 and Volume 33, Number 2, Pages 443 - 445, 1985) or the Lotter algorithm (see T. Lotter, P. Vary, "Noise Reduction by Joint Maximum A Posteriori Spectral Amplitude And Phase Estimation with Super-Gaussian Speech Modelling", EUSIPCO '04, Conference Proceedings, Number 2, Pages 1457 - 1460) may be employed as well.

[0064] With the above described method, an initial target noise $S_{bb,target}(\omega_{\mu})$ as measured in a first vehicle may be used. If this initial target noise is then employed in a different vehicle, the residual noise of this different vehicle is matched to the residual noise of the first vehicle in a level adjusted way.

[0065] The disclosed method has the additional advantage that non-stationary noise can be dealt with in an improved way. In the upper-part of Figure 2, a time-frequency analysis of a microphone signal is shown. This analysis corresponds to the noise in a vehicle at a speed of 100 km/h. After about two seconds, another vehicle is approaching resulting in additional noise as indicated by the elliptic frame.

[0066] In the middle part of Figure 2, the time-frequency analysis of a conventional noise reduction method is shown. As one can see, only part of the non-stationary noise has been removed. For this conventional noise reduction method, the following Wiener characteristic was used:

$$G(e^{j\Omega_{\mu}}, n) = \max \left\{ G_{\min}, 1 - \beta(e^{j\Omega_{\mu}}, n) \frac{\hat{S}_{bb}(\Omega_{\mu}, n)}{\hat{S}_{yy}(\Omega_{\mu}, n)} \right\},$$

wherein G_{\min} is constant and equal to 0.3.

[0067] In the lower part of Figure 2, the above-described method according to the present invention has been applied resulting in an almost complete removal of this non-stationary noise.

[0068] A further advantage is illustrated in Figure 3. In the upper-part of this Figure, a tonal disturbance at about 3,000 Hz is present in a microphone signal. A conventional noise reduction method slightly reduces this noise by about 10 to 15 dB (see the middle part of Figure 3). In contrast to this, the method according to the present invention removes this tonal noise almost completely.

[0069] In the illustrated embodiments described above, a single target noise spectrum is used. It is to be understood that more than one target noise spectrum may be used as well. For example, a first target noise spectrum may be provided for small velocities of a vehicle, a second target noise spectrum for medium velocities and a third target noise spectrum for high velocities. Depending on the current speed of the vehicle, the noise reduction system may switch from one target noise spectrum to the other.

[0070] It is to be understood that the different parts and components of the method and apparatus described above can also be implemented independent of each other and be combined in different forms. Furthermore, the above-described embodiments are to be construed as exemplary embodiments only.

Claims

1. Method for processing an acoustic input signal to provide an output signal with reduced noise, comprising weighting the input signal using a frequency dependent weighting function, wherein the weighting function is bounded below by a frequency dependent threshold function.
2. Method according to claim 1, wherein the threshold function is a time dependent function.
3. Method according to one of the preceding claims, comprising performing wanted signal detection and adapting the weighting function if no wanted signal is detected.
4. Method according to one of the preceding claims, wherein the threshold function is based on a target noise spectrum.
5. Method according to claim 4, wherein the target noise spectrum is time dependent.
6. Method according to claim 4 or 5, comprising performing wanted signal detection and adapting the target noise spectrum if no wanted signal is detected.

- 5
7. Method according claim 6, wherein the target noise spectrum at time n is incremented if the power of the target noise spectrum at time $(n-1)$ within a predetermined frequency interval is smaller than a predetermined attenuation factor times an estimate of the power of a noise component in the input signal at time n within the predetermined frequency interval.
- 10
8. Method according to one of the claims 4 to 7, wherein the threshold function is based on the minimum of a predetermined minimum attenuation value and a quotient of the target noise spectrum and the absolute value of the input signal.
- 15
9. Method according to claim 8, wherein the threshold function is based on the maximum of said minimum and a predetermined maximum attenuation value.
10. Method according to claim 9, wherein the threshold function at time n is based on a convex combination of the threshold function at time $(n-1)$ and said maximum at time n .
- 20
11. Method according to one of the claims 4 to 10, wherein the threshold function is based on at least two target noise spectra.
12. Method according to one of the preceding claims, wherein the weighting function is based on an estimated power density spectrum of a noise signal component and/or an estimated power density spectrum of the input signal.
- 25
13. Method according to one of the preceding claims, wherein the weighting function is based on the maximum of the threshold function and a predetermined filter characteristic.
14. Method according to claim 12, wherein the filter characteristic is time dependent.
- 30
15. Method according to one of the preceding claims, wherein the weighting function is based on a Wiener characteristic.
16. Method according to one of the preceding claims, wherein the method is performed in the frequency domain.
- 35
17. Method according to claim 16, wherein at least one of the steps is performed for each frequency subband separately.
18. Computer program product comprising one or more computer readable media having computer-executable instructions for performing the steps of the method of one of the preceding claims when run on a computer.
- 40
19. Apparatus for processing an acoustic input signal to provide an output signal with reduced noise, comprising means for weighting the input signal using a frequency dependent weighting function, wherein the weighting function is bounded below by a frequency dependent threshold function.
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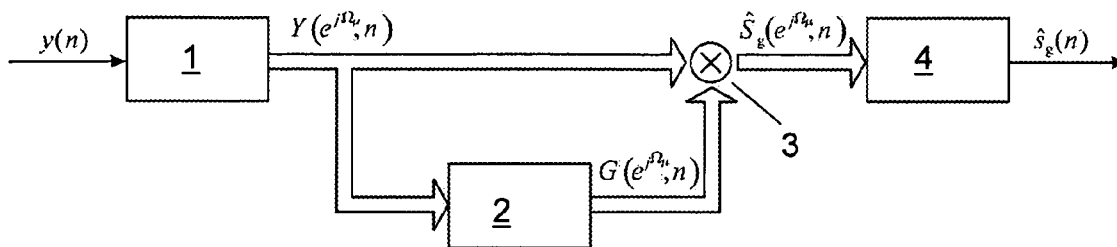


Fig. 1

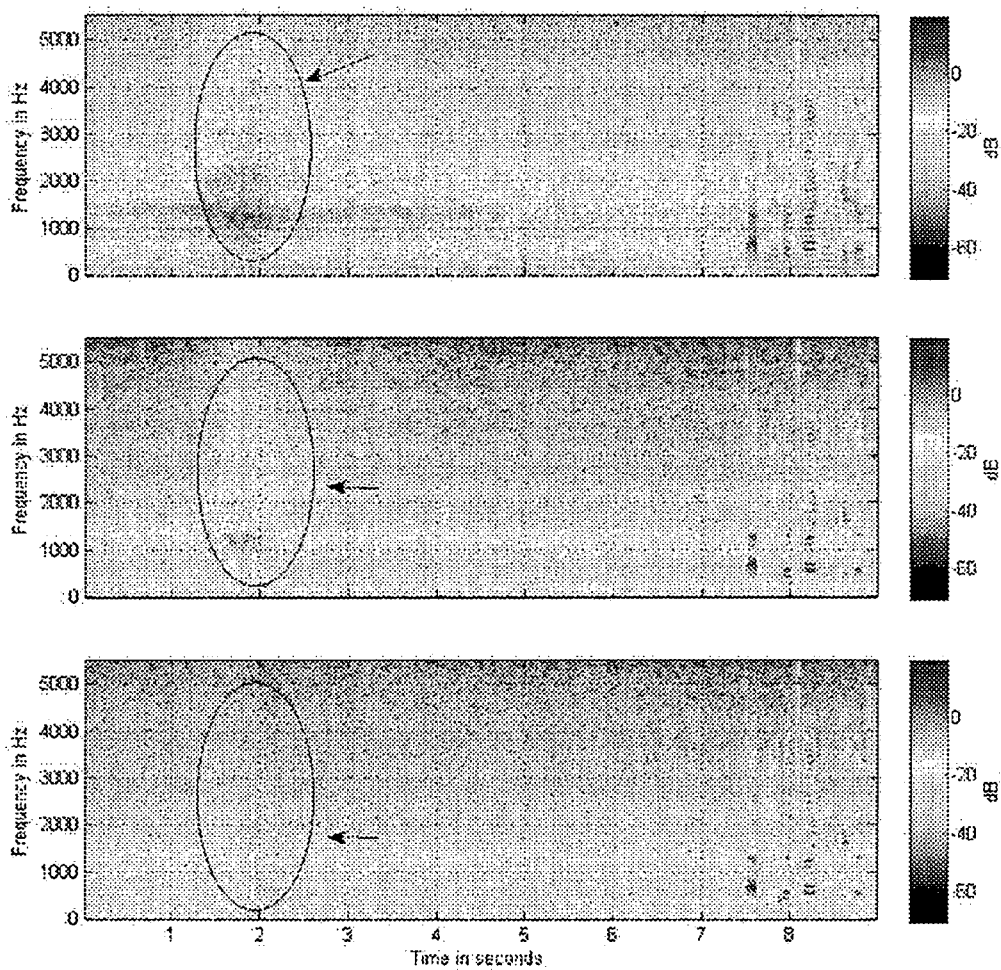


Fig. 2

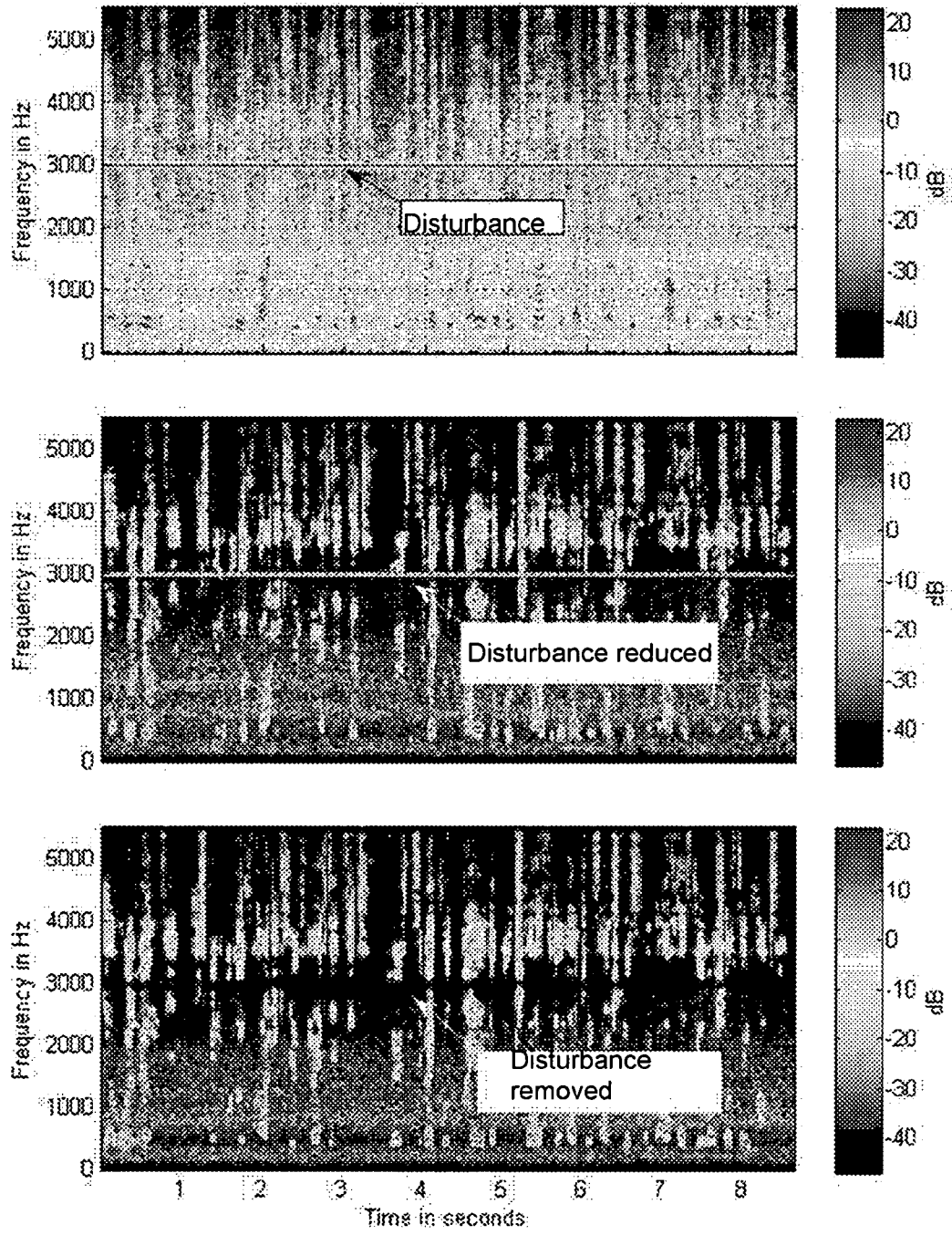


Fig. 3



DOCUMENTS CONSIDERED TO BE RELEVANT			
Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (IPC)
A	WO 01/13364 A (WAVEMAKERS RES INC [CA]; ZAKARAUSKAS PIERRE [US]) 22 February 2001 (2001-02-22) * abstract * * page 7, lines 8-10 * * page 11, lines 12-23 * -----	1-19	INV. G10L21/02
A	US 2003/128851 A1 (FURUTA SATORU [JP]) 10 July 2003 (2003-07-10) * abstract; figures 7-9 * * paragraphs [0081] - [0087] * -----	1-19	
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A	US 6 453 289 B1 (ERTEM FILIZ BASBUG [US] ET AL) 17 September 2002 (2002-09-17) * abstract * * column 9, line 1 - column 10, line 15 * -----	1-19	
The present search report has been drawn up for all claims			TECHNICAL FIELDS SEARCHED (IPC) G10L
Place of search The Hague		Date of completion of the search 25 October 2007	Examiner Quélavoine, Régis
CATEGORY OF CITED DOCUMENTS X : particularly relevant if taken alone Y : particularly relevant if combined with another document of the same category A : technological background O : non-written disclosure P : intermediate document		T : theory or principle underlying the invention E : earlier patent document, but published on, or after the filing date D : document cited in the application L : document cited for other reasons & : member of the same patent family, corresponding document	

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EPO FORM 1503 03 82 (P04C01)

**ANNEX TO THE EUROPEAN SEARCH REPORT
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