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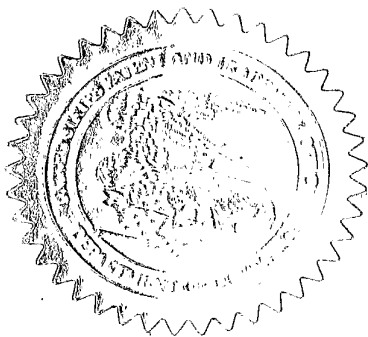
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PROVISIONAL APPLICATION FOR PATENT COVER SHEET - Page 1 of 2
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INVENTOR(S)		
Given Name (first and middle (if any))	Family Name or Surname	Residence (City and either State or Foreign Country)
Rob Anton Jurjen	DEVRIES	Tilburg, The Netherlands

Additional inventors are being named on the _____ separately numbered sheets attached hereto

TITLE OF THE INVENTION (500 characters max):

A HEARING INSTRUMENT WITH ADAPTIVE DIRECTIONAL SIGNAL PROCESSING

Direct all correspondence to: CORRESPONDENCE ADDRESS

☒ The address corresponding to Customer Number: 23639

OR

☐ Firm or Individual Name **Bingham McCutchen LLP**

Address **Three Embarcadero Center, 18th Floor**

City San Francisco	State CA	Zip 94111
Country USA	Telephone 650-849-4400	Email gerald.chan@bingham.com

ENCLOSED APPLICATION PARTS (check all that apply)

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| ☐ Application Data Sheet. See 37 CFR 1.76 | ☐ CD(s), Number of CDs _____ |
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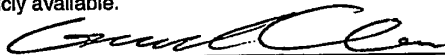
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SIGNATURE



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TYPED or PRINTED NAME Gerald Chan

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TELEPHONE 650-849-4960

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A HEARING INSTRUMENT WITH ADAPTIVE DIRECTIONAL SIGNAL PROCESSING FIELD

The present application relates to a hearing instrument, such as a hearing aid, an implantable hearing prosthesis, a head set, a mobile phone, etc, with a signal
5 processor that is adapted for directional signal processing.

BACKGROUND

It is well-known to use information on the directions to sound sources in relation to a listener for distinguishing between noise sources and desired sound sources.
Throughout the present specification, the term directional signal processing system
10 means a signal processing system that is adapted to exploit the spatial properties of an acoustic environment. Directional microphones are available, but typically directional signal processing systems utilize an array of omni-directional microphones.

The directional signal processing system combines the electrical signals from the microphones in the array into a signal with varying sensitivity to sound sources in
15 different directions in relation to the array. Throughout the present specification, a plot of the varying sensitivity as a function of the direction is denoted the directivity pattern. Typically, a directivity pattern has at least one direction wherein the microphone signals substantially cancel each other. Throughout the present specification, such a direction is denoted a null direction. A directivity pattern may comprise several null directions
20 depending on the number of microphones in the array and depending on the signal processing.

Directional signal processing systems are known that prevent sound suppression of sources in certain directions of interest.

For example, US 5,473,701 discloses a method of enhancing the signal-to-noise ratio
25 of a microphone array with an adjustable directivity pattern, i.e. an adjustable null direction, for reduction of the microphone array output signal level in accordance with a criterion wherein the reduction is performed under a constraint that the null direction is precluded from being located within a predetermined region of space.

SUMMARY

30 It is an object of the present application to provide a system with an improved capability of suppressing sound sources from all directions.

According to some embodiments, the above-mentioned and other objects are fulfilled by a hearing instrument with at least two microphones for reception of sound and conversion of the received sound into corresponding electrical sound signals that are input to a signal processor, wherein the signal processor is adapted to process the electrical sound signals into a combined signal with a directivity pattern with at least one adaptive null direction θ . The signal processor is further adapted to prevent the at least one adaptive null direction θ from entering a prohibited range of directions, wherein the prohibited range is a function of a parameter of the electrical sound signals.

- 10 Preferably, the at least two microphones are omni-directional microphones; however in some embodiments, some of the at least two microphones are substituted with directional microphones.

It is an important advantage that suppression of desired sound sources are avoided while undesired sound sources may still be suppressed from any arbitrary direction.

- 15 The hearing instrument may further comprise a speech detector for detection of presence of speech. Adjustment of the prohibited range of directions may be performed gradually over a first time interval when speech is detected after a period of absence of speech.

- 20 Further, adjustment of the prohibited range of directions may be performed gradually over a second time interval when speech stops after a period of presence of speech.

The prohibited range may include a predetermined direction, such as 0° azimuth.

An estimate of the power of sound received by at least one of the at least two microphones may constitute the parameter, for example the averaged power of sound received by a front microphone may constitute the parameter.

- 25 An estimate of the signal to noise ratio of sound received by at least one of the at least two microphones may constitute the parameter.

The hearing instrument may further comprise a speech detector and a direction of arrival detector, and the prohibited range may include the detected direction of arrival of detected speech whereby suppression of the speech is prevented.

- 30 The prohibited range may, in the presence of multiple speech sources, include the detected direction of arrival of detected speech closest to 0° azimuth.

The prohibited range may, in the presence of multiple speech sources, include the detected directions of arrival of all speech sources.

A current null direction may reside inside the prohibited range of directions upon adjustment of the prohibited range of directions. The signal processor may further be adapted to move such a null direction outside the adjusted prohibited range.

The signal processor may be adapted for subband processing whereby

the electrical sound signals from the microphones are divided into a set of frequency bands B_i , and, in each frequency band B_i , or at least in some of the frequency bands B_i , the electrical sound signals are individually processed including

processing the electrical signals into a combined signal with an individual directivity pattern with an individually adapted null direction θ_i and

preventing the null direction θ_i from entering a prohibited range of directions, wherein the prohibited range is a function of a parameter of the electrical sound signals.

Subband processing allows individual suppression of undesired sound sources emitting sound in different frequency ranges.

The signal processor may be adapted to perform directional signal processing selected from the group consisting of an adaptive beam former, a multi-channel Wiener filter, an independent component analysis, and a blind source separation algorithm.

BRIEF DESCRIPTION OF THE DRAWINGS

The above and other features and advantages will become more apparent to those of ordinary skill in the art by describing in detail exemplary embodiments thereof with reference to the attached drawings in which:

Fig. 1 shows a simplified block diagram of a digital hearing aid, and

Fig. 2 schematically illustrates the directional signal processing of the hearing aid of

Fig. 1.

DETAILED DESCRIPTION OF PREFERRED EMBODIMENTS

The present embodiments will now be described more fully hereinafter with reference to the accompanying drawings, in which exemplary embodiments are shown. The embodiments may, however, be embodied in different forms and should not be construed as limited to the embodiments set forth herein. Like reference numerals refer to like elements throughout.

Fig. 1 shows a simplified block diagram of a digital hearing aid. The hearing aid 1 comprises one or more sound receivers 2, e.g. two microphones 2a and a telecoil 2b. The analog signals for the microphones are coupled to an analog-digital converter circuit 3, which contains an analog-digital converter 4 for each of the microphones.

- 5 The digital signal outputs from the analog-digital converters 4 are coupled to a common data line 5, which leads the signals to a digital signal processor (DSP) 6. The DSP is programmed to perform the necessary signal processing operations of digital signals to compensate hearing loss in accordance with the needs of the user. The DSP is further programmed for automatic adjustment of signal processing parameters.
- 10 The output signal is then fed to a digital-analog converter 12, from which analog output signals are fed to a sound transducer 13, such as a miniature loudspeaker.

In addition, externally in relation to the DSP 6, the hearing aid contains a storage unit 14, which in the example shown is an EEPROM (electronically erasable programmable read-only memory). This external memory 14, which is connected to a common serial data bus 17, can be provided via an interface 15 with programmes, data, parameters etc. entered from a PC 16, for example, when a new hearing aid is allotted to a specific user, where the hearing aid is adjusted for precisely this user, or when a user has his hearing aid updated and/or re-adjusted to the user's actual hearing loss, e.g. by an audiologist.

- 20 The DSP 6 contains a central processor (CPU) 7 and a number of internal storage units 8-11, these storage units containing data and programmes, which are presently being executed in the DSP circuit 6. The DSP 6 contains a programme-ROM (read-only memory) 8, a data-ROM 9, a programme-RAM (random access memory) 10 and a data-RAM 11. The two first-mentioned contain programmes and data which constitute
- 25 permanent elements in the circuit, while the two last-mentioned contain programmes and data which can be changed or overwritten.

Typically, the external EEPROM 14 is considerably larger, e.g. 4-8 times larger, than the internal RAM, which means that certain data and programmes can be stored in the EEPROM so that they can be read into the internal RAMs for execution as required.

- 30 Later, these special data and programmes may be overwritten by the normal operational data and working programmes. The external EEPROM can thus contain a series of programmes, which are used only in special cases, such as e.g. start-up programmes.

Fig. 2 schematically illustrates the signal processing of a hearing instrument according to some embodiments. The illustrated hearing instrument has two microphones 20, 22 positioned in a housing to be worn at the ear of the user. When the housing is mounted in its operating position at the ear of the user, one of the microphones, the front microphone 20, is positioned in front of the other microphone, the rear microphone 22, and a horizontal line extending through the front and rear microphones defines the front direction, i.e. azimuth = 0° , corresponding to the looking direction of the user of the hearing instrument.

In another embodiment comprising a binaural hearing aid, the microphones 20, 22 may be positioned in separate housing, namely a housing positioned in the left ear and a housing positioned in the right ear of the user. The directional signal processing may take place in either of the left or right hearing aid housings, or in both housing, or in a separate housing containing signal processing circuitry and intended to be worn elsewhere on the body of the user. The electrical signals may be communicated between the housings with electrical wires or wirelessly. The large distance between microphones in the left ear housing and the right ear housing may lead to a directivity pattern with a large directivity.

The microphones 20, 22 convert received sound signals into corresponding electrical sound signals that are converted into digital sound signals 24, 26 by respective A/D converters (not shown).

Each of the digitized sound signals 24, 26 is input to a respective subtraction circuit 28, 30 and a respective delay 32, 34 with delay D_H . Each delay 32, 34 delays the digitized sound signal 24, 26 by the amount of time used by a sound signal to propagate in the 0° azimuth direction from the front microphone 20 to the rear microphone 22. Each subtraction circuit 28, 30 subtracts the respective delayed signal 36, 38 from one microphone 20, 22 from the direct signal 26, 24 of the other microphone 22, 20. Each of the subtracted signals 40, 42 has a fixed directional pattern 44, 46, a so-called cardioid pattern. The cardioid pattern 44 of the upper branch (a) has a null direction 48 at 180° azimuth, i.e. pointing in the rear direction of the user, and the cardioid pattern 46 of the lower branch (b) has a null direction 50 at 0° azimuth, i.e. pointing in the front direction of the user.

The subtracted signal 42 of the lower branch (b) is filtered by an adaptive filter 52 with a transfer function H , and the subtracted signal 40 of the upper branch (a) is delayed by a delay 54 with a delay D_H equal to the group delay of the adaptive filter 52, and subsequently the two signals 56, 58 are subtracted for formation of a combined signal

60 with a directivity pattern 62 with an adaptive null direction θ . An example of a resulting directivity pattern 62 is also shown in Fig. 2. The hatched area of the resulting directivity pattern 62 illustrates the prohibited range of directions which in the illustrated example is symmetrical around 0° azimuth. The arched arrows indicate that the prohibited range of directions vary as a function of a parameter of the electrical sound signals.

It should be noted that in the illustrated embodiment of Fig. 2, the delay 34 and the subtraction circuit 28 may be omitted and still an output 60 with a directional pattern 62 similar to the illustrated embodiment of Fig. 2 may be obtained due to corresponding changes in the operation of the adaptive filter 52.

Further, both delays 32, 34 and subtraction circuits 28, 30 may be omitted in the illustrated embodiment of Fig. 2, and still an output 60 with a directional pattern 62 similar to the illustrated embodiment of Fig. 2 may be obtained due to corresponding changes in the operation of the adaptive filter 52.

In the illustrated embodiment, the filter 52 is adapted to minimize the output power of the combined signal 60 by the filter coefficient update circuit 64. The filter 52 may be a finite impulse response (FIR) filter with N taps.

The adaptive filter controller 66 prevents the null direction θ from entering a prohibited range of directions as a function of a parameter of the electrical sound signals.

The adaptive filter controller 66 constrains the filter coefficients of the adaptive filter 52 in such a way that a directional null θ remains outside the prohibited range of directions.

For example, the adaptive filter 52 may have a single tap in which case the adaptive filter 52 is an amplifier with a gain G_H , and the adaptive filter controller 66 constrains the gain G_H to remain inside the range $0 \leq G_H \leq G_{\text{limit}}$. The value of the threshold G_{limit} determines the prohibited range of directions. For example, when $G_{\text{limit}} = 1$, the prohibited range of directions ranges from -90° azimuth to $+90^\circ$ azimuth.

The adaptive filter controller 66 may freeze the filter coefficients, i.e. updating of the filter coefficients may be stopped temporarily, when the strongest sound source is located within the prohibited range of directions. This approach requires estimation of the direction of arrival (DOA) of the signal incident on the hearing instrument.

A DOA estimate may be obtained by determination of an M point auto-correlation A of the front microphone signal 24 delayed by D and determination of an M point cross-

correlation B of the front microphone signal 24 delayed by D and the rear microphone signal 26:

$$A = \sum_{i=0}^{M-1} (\text{front}(k - D - i))^2 \quad (1)$$

$$B = \sum_{i=0}^{M-1} \text{front}(k - D - i) \text{rear}(k - i) \quad (2)$$

$\beta = B/A$ can be used as an estimate of the direction of arrival of the dominant sound in the acoustic environment. When $\beta = B/A = 1$, the DOA is 0° . As β decreases toward 0, the DOA moves towards 180° azimuth. Thus, the adaptation may be temporarily stopped when

$$\beta > \sigma \quad (3)$$

where σ is determined in such a way that $\beta = B/A = \sigma$ when the DOA of e.g. a zero mean white noise source is α degrees azimuth, the prohibited range of directions extending from $-\alpha$ degrees azimuth to α degrees azimuth including 0° azimuth.

It should be noted that with this DOA estimate, the prohibited range of directions will be frequency dependent, because the value of $\beta = B/A$ is both dependent on the direction of arrival and on the frequency of the signal. In an embodiment with subband processing with individual beamforming in each frequency band B_i , individual thresholds σ_i may be defined for each frequency band B_i .

The person skilled in the art will recognize that numerous other conventional methods are available to obtain an estimate of the DOA, including frequency independent estimates.

The signal processing is not necessarily done on the same apparatus that contains (one or more of) the microphones. The signal processing may be performed in a separate device that is linked to the, possibly multiple apparatuses that contain the microphones via a wire, wireless or other connection.

In the following various examples are described of determining the prohibited range of directions as a function of a parameter of the electrical sound signals. In the examples, $-\alpha$ till α degrees azimuth constitutes the prohibited range of directions including 0° azimuth.

In one embodiment, the prohibited range of directions is a function of the short term average power P_F (e.g. over the past 10 seconds) of the electrical signal 24 from front microphone 20 in accordance with

$$\alpha = \alpha_{\max} \left(1 - \frac{\max(P_{\min}, \min(P_F, P_{\max})) - P_{\min}}{P_{\max} - P_{\min}} \right) \quad (4)$$

- 5 Hence, the prohibited range of directions narrows when the signal power P_F increases and for $P_F > P_{\max}$ $\alpha = 0^\circ$ (front direction) and for $P_F < P_{\min}$ $\alpha = \alpha_{\max}$.

The values of $\alpha_{\max}$, $P_{\min}$, and $P_{\max}$ may be set during manufacture of the hearing instrument, or, during a fitting session of the hearing instrument with the intended user.

- 10 In one example, $P_{\min} = 45 \text{ dB}_{\text{SPL}}$ and $P_{\max} = 110 \text{ dB}_{\text{SPL}}$. It should be noted that very loud sounds (above $110 \text{ dB}_{\text{SPL}}$) may be suppressed from any direction providing protection against harmful sounds (e.g. when getting too close to a loudspeaker at a concert). For $\alpha_{\max} = 180^\circ$, an omni-directional pattern is obtained in relative quiet environments below $45 \text{ dB}_{\text{SPL}}$.

- 15 In another embodiment, the prohibited range of directions is a function of the signal-to-noise ratio SNR for the signal 40 at point (a) in Fig. 1 in accordance with

$$\alpha = \alpha_{\max} \frac{\max(\text{SNR}_{\min}, \min(\text{SNR}, \text{SNR}_{\max})) - \text{SNR}_{\min}}{\text{SNR}_{\max} - \text{SNR}_{\min}} \quad (5)$$

Hence, the prohibited range of directions narrows when the signal-to-noise ratio SNR increases and for $\text{SNR} > \text{SNR}_{\max}$ $\alpha = 0^\circ$ (front direction) and for $\text{SNR} < \text{SNR}_{\min}$ $\alpha = \alpha_{\max}$.

- 20 The values of $\alpha_{\max}$, $\text{SNR}_{\min}$, and $\text{SNR}_{\max}$ may be set during manufacture of the hearing instrument, or, during a fitting session of the hearing instrument to the intended user.

- 25 SNR may be estimated utilizing a speech detector 68, e.g. a modulation or speech probability estimator, or a modulation or speech activity detector, to detect presence of speech and calculate the average power P_X of the signal when speech is present. The average noise power P_N in absence of speech is estimated using a minimum statistics approach. An estimate of the SNR is then given by

$$\text{SNR} = 20 \log_{10} \left(\frac{P_X - P_N}{P_N} \right) \text{ dB} \quad (6)$$

In one embodiment, the prohibited range of directions is a function of the azimuth direction of speech β . Presence of speech is detected by the speech detector 68 that

processes the signal 24 and the direction of arrival β is estimated by the direction of arrival detector 70, and the prohibited range of directions is adjusted to include β . β may change due to head or speaker movement. In the presence of multiple speech sources, the prohibited range of directions may be adjusted to include DOA of the speech source closest to 0° azimuth or to include DOAs of all detected speech sources.

The above-mentioned approaches may be combined.

For example in one embodiment, the prohibited range of directions is a function of the short term average power P_F (e.g. over the past 10 seconds) of the electrical signal 24 from front microphone 20 in accordance with

$$\alpha = \alpha_{\max} - \frac{\max(P_{\min}, \min(P_F, P_{\max})) - P_{\min}}{P_{\max} - P_{\min}} (\alpha_{\max} - \alpha_{\text{snr}}) \quad (7)$$

which is similar to equation (4) above with the exception that α varies between $\alpha_{\max}$ and α_{snr} in equation (7) while α varies between $\alpha_{\max}$ and 0° in equation (4), and wherein

$$\begin{aligned} \alpha_{\text{snr}} = \alpha_{\min} &+ \frac{\max(\text{SNR}_{\min}, \min(\text{SNR}, \text{SNR}_{\max})) - \text{SNR}_{\min}}{\text{SNR}_{\max} - \text{SNR}_{\min}} (180 - \alpha_{\min}) \\ &+ \frac{\max(\text{SNR}_{\text{lowthld}}, \min(\text{SNR}, \text{SNR}_{\text{low}})) - \text{SNR}_{\text{low}}}{\text{SNR}_{\text{lowthld}} - \text{SNR}_{\text{low}}} (180 - \alpha_{\min}) \end{aligned} \quad (8)$$

and

$$\alpha_{\min} = \text{DOA}_{\max} \quad (9)$$

wherein

SNR is the estimated signal-to-noise ratio at point (a) in Fig. 1 over the past 10 seconds, e.g. obtained as described above,

SNR_{short} is the estimated signal-to-noise ratio at point (a) in Fig. 1 over the past 0.05 seconds, e.g. obtained as described above,

SNR_{shortmax} is the maximum value of SNR_{short} over the past 10 seconds, and

DOA_{max} is the average value of the DOA over the 0.05 seconds block that resulted in SNR_{shortmax}.

For example, $P_{\max} = 60 \text{ dB}_{\text{SPL}}$, $P_{\min} = 45 \text{ dB}_{\text{SPL}}$, $\text{SNR}_{\min} = 5 \text{ dB}$, $\text{SNR}_{\max} = 15 \text{ dB}$, $\text{SNR}_{\text{low}} = -10 \text{ dB}$, $\text{SNR}_{\text{lowthld}} = -20 \text{ dB}$, and $\alpha_{\max} = 180^\circ$.

It should be noted that in this embodiment the prohibited range of directions is as narrow as possible around the direction to the speech source with the highest SNR.

- 5 The prohibited range increases when the overall SNR is larger than the threshold $\text{SNR}_{\min}$ or smaller than the threshold SNR_{low} , and saturates into an omni-directional pattern when the SNR is larger than the threshold $\text{SNR}_{\max}$ (e.g. when there is no noise) or lower than the threshold $\text{SNR}_{\text{lowthld}}$ (e.g. when there is no speech), or the overall signal power P_F is smaller than the threshold $P_{\max}$, and also saturates into an omni-
10 directional pattern when P_F is smaller than the threshold $P_{\min}$ (e.g. in quiet surroundings).

Preferably, timing restrictions are also included so that frequent and abrupt changes of the prohibited range of directions are avoided.

- For example the prohibited range of directions may be prevented from narrowing in
15 response to a short term presence of a noise source, such as reception of reverberations. Short term presence may be defined as presence during less than 0.1 seconds.

- An adjustment of the prohibited range of directions may be performed gradually in a time interval when speech stops after a period of presence of speech. For example, α
20 may be gradually increased to $\alpha_{\max}$ in a time interval of about 3 seconds. Throughout the present specification, presence or absence of speech refer to the detection or non-detection of speech, respectively, of the system.

- A speech stop may be defined as the moment that no speech has been detected for e.g., 5 seconds, and a conversation stop may be defined as the moment that no
25 speech has been detected for e.g., 30 seconds. Speech start and conversation start may be defined as the moment that speech is detected for the first time after a speech stop and a conversation stop, respectively.

A long term average may be defined as the average over e.g., 2 seconds. A short term average may be defined as the average over e.g., 50 milliseconds.

- 30 In one embodiment, the prohibited range of directions is adjusted upon start of conversation according to the following:

Calculation of the long term average DOA value during speech presence is performed; typically the calculation requires 2 seconds of speech presence.

- Provided that the long term average DOA value during speech presence is not significantly different from the long term average DOA value during speech absence, α is increased to $\alpha_{\max}$ with the release time, e.g. in about 3 seconds. (This situation occurs when e.g. the noise and speech arrive from the same direction, in which case
- 5 beamforming is not advantageous, or when the speaker is outside the Hall radius and the perceived noise field is diffuse, or when the SNR is low.)

Provided that the long term average DOA value during speech presence is significantly different from the long term average DOA value during speech absence, the prohibited range of direction is adjusted in accordance with the following:

- 10 When the short term average DOA value during speech presence remains above or around e.g. 80° , α is increased to $\alpha_{\max}$ in about 3 seconds. (In this case the listener is apparently not interested enough in the speech to turn his head, or he is e.g. driving a car and can not turn his head to the speaker.)

- 15 When the short term average DOA value during speech presence does become significantly lower than 80° , the prohibited range of directions is adjusted to just include the minimum of the short term average DOA value over e.g. the past 2 seconds, plus a safety margin of about 20° in order to take head movements into account. This is repeated until speech stop. Upon speech stop, α is adjusted to e.g. $\phi_{\max} + 20^\circ$, where $\phi_{\max}$ is equal to the maximum of the short term average DOA values measured at a
- 20 speech start over e.g. the last 3 speech start events. (This prevents the user from missing any of the speech of interest, while a narrow beam is also obtained when the user has focused on the speaker. A situation like this can occur when the user is in a restaurant and is alternatively looking at the plate and at the person next or opposite to the user.)

- 25 In the above example, preferably $\alpha_{\max}$ is 180° so that an omni-directional pattern is obtained when α is increased to $\alpha_{\max}$, since the omni-directional pattern imparts a perception to the user of being connected to the environment.

$\alpha_{\max}$ equal to 90° may be selected to maintain directional suppression in the back region of the user.

- 30 The prohibited range of directions may be broadened to such an extent that an existing null direction θ ends up residing within the prohibited range. In a preferred embodiment, a null direction θ residing within the prohibited range for a certain time period, e.g. 10 seconds, is moved outside the prohibited range. This may be done momentarily or over a period of time.

For example, the adaptive filter may be forced to position the null direction at $\theta = 180^\circ$ and continue adaptation from this value. E.g., the coefficients of the adaptive filter 52 may be reset to values that position the null direction at $\theta = 180^\circ$. In another example wherein the cost function that H minimizes is equal to the output power, a weighted
 5 bias term is added to the cost function that forces H to position the directional null at 180° . When the null has moved outside the prohibited range of directions, the normal adaptation is resumed.

A value of α to 180° indicates that an omni-directional pattern is desired. An omni-directional pattern may be obtained by processing the output 66 in accordance with:

$$10 \quad z(k) = y(k) + (1 - \lambda) (x(k) - y(k)) \quad (10)$$

wherein y is the output 60, x is the output 24 of the front microphone 20, and k is the current sample number. z is the processed output.

Hence, when λ equals zero, z is equal to x, i.e. an omni-directional pattern is provided, and when λ equals 1, z is equal to the original directional output 60. By varying λ
 15 between 0 and 1, the directional pattern changes gradually from the original directional output 60 to an omni-directional output. Thus, when α gets equal to 180° , λ is decreased to 0 in e.g. 10 seconds and when α becomes smaller than 180° λ is increased towards 1 in e.g. 3 seconds.

CLAIMS

1. A hearing instrument with at least two microphones for reception of sound and conversion of the received sound into corresponding electrical sound signals that are input to a signal processor, wherein the signal processor is adapted to process the electrical sound signals into a combined signal with a directivity pattern with at least one adaptive null direction θ ,
 5 characterized in that
 the signal processor is further adapted to prevent the at least one null direction θ from entering a prohibited range of directions, wherein the prohibited range is a function of a parameter of the electrical sound signals.
 10
2. A hearing instrument according to claim 1, further comprising a speech detector, and wherein adjustment of the prohibited range of directions is performed gradually in a time interval when speech is detected after a period of absence of speech.
3. A hearing instrument according to claim 1 or 2, further comprising a speech detector and wherein adjustment of the prohibited range of directions is performed gradually in a time interval when speech stops after a period of presence of speech.
 15
4. A hearing instrument according to any of the previous claims, wherein the prohibited range includes a predetermined direction.
5. A hearing instrument according to any of the previous claims, wherein a power estimate of sound received by at least one of the at least two microphones constitutes the parameter.
 20
6. A hearing instrument according to any of the previous claims, wherein an estimate of the signal to noise ratio of sound received by at least one of the at least two microphones constitutes the parameter.
7. A hearing instrument according to any of the previous claims, further comprising a speech detector and a direction of arrival detector and wherein the prohibited range includes the detected direction of arrival of detected speech.
 25
8. A hearing instrument according to claim 7, wherein the prohibited range, in the presence of multiple speech sources, includes the detected direction of arrival of detected speech closest to 0° azimuth.
 30

9. A hearing instrument according to claim 7, wherein the prohibited range, in the presence of multiple speech sources, includes the detected directions of arrival of all speech sources.

10. A hearing instrument according to any of the previous claims, wherein the signal processor is further adapted to move a current null direction outside an adjusted prohibited range.

11. A hearing instrument according to any of the previous claims, wherein the signal processor is adapted for subband processing whereby

the electrical sound signals from the microphones are divided into a set of frequency bands B_i , and, in each frequency band B_i , the electrical sound signals are individually processed including

in at least one subband, processing the electrical signals into a combined signal with an individual directivity pattern with an individually adapted null direction θ_i , and

preventing the null direction θ_i from entering a prohibited range of directions, wherein the prohibited range is a function of a parameter of the electrical sound signals.

12. A hearing instrument according to any of the previous claims, wherein the signal processor is adapted to perform directional signal processing selected from the group consisting of an adaptive beam former, a multi-channel Wiener filter, an independent component analysis, and a blind source separation algorithm.

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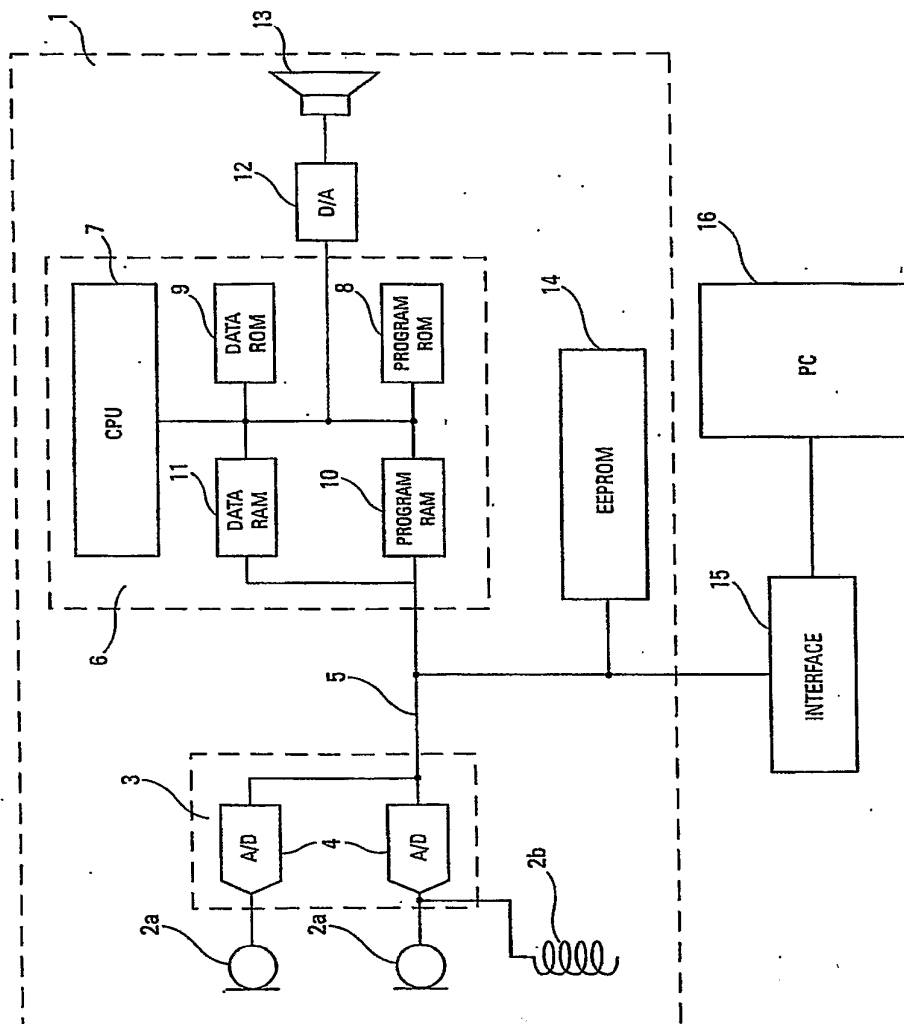


Fig. 1

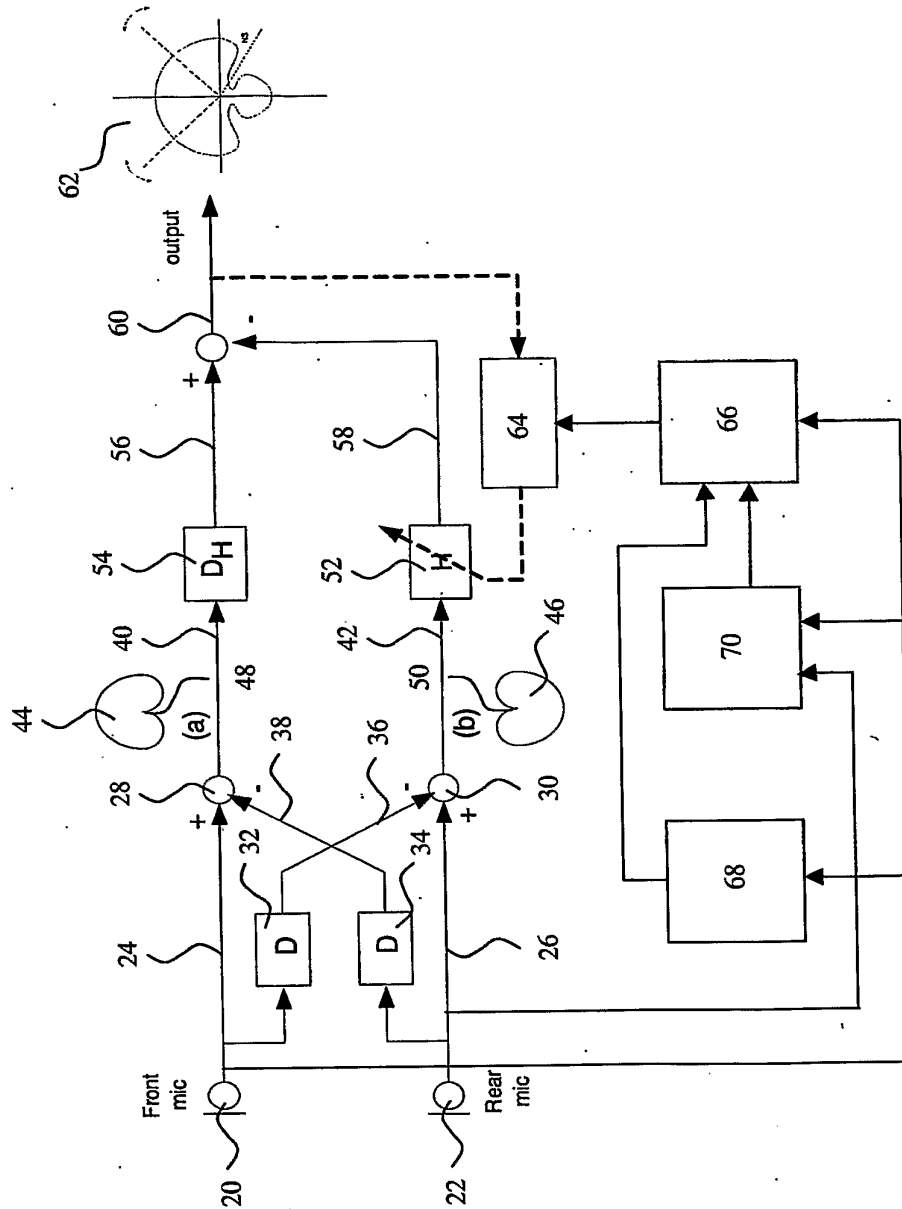


Fig. 2