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(54) **Automatic microphone matching**

(57) Matching of at least two acoustical to electrical converters is performed in dependency of an impinging direction of arrival within a range determined before matching. The two acoustical to electrical converters provide as an example beamforming ability for a cardoid transfer characteristic (c). Matching of the two acoustical/electrical converters involved is performed whenever direction of arrival (DOA) of acoustical signals impinging on the converter is within a range $DOA_S \pm \Delta DOA$. Such range is selected around a direction of arrival where e.g. the transfer characteristic is zero (d).

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Description

[0001] This application has an Attachment A

[0002] The present invention is directed on a method for matching at least two acoustical to electrical converters which generate, respectively, electrical output signals. Signals which depend on the electrical output signals of the converters are computed to result in a result signal. The transfer characteristic between an acoustical signal impinging upon the at least two converters and the result signal is dependent on direction of arrival - DOA - of the acoustical signal upon the at least two converters.

[0003] Acoustical pickup arrangements which have a transfer characteristic between acoustical input and electrical output, the amplification thereof being dependent on the DOA of acoustical signals on the acoustical inputs of such devices are called "beamformers" and are widely used as e.g. for hearing devices, be it outside-the-ear hearing devices or in-the-ear hearing devices, be it for such hearing devices to improve and facilitate normal hearing or be it for such hearing devices for therapeutic appliances, i.e. to improve hearing capability of hearing impaired persons. Further, beamformers may also be applied for hearing protection devices, whereat the main target is to protect an individual from excessive acoustical loads.

[0004] The addressed transfer characteristic, called the "beam" characteristic when represented in polar coordinates, is of one or more than one lobe and has accordingly one or more minima, called "Nulls", at specific values of DOA.

[0005] Beamformers may be conceived just by acoustical to electrical converters which per se have a beamforming characteristic.

[0006] The present invention deals with other cases where at least two spaced apart acoustical to electrical converters are used, signals dependent on their electrical output signals being computed to generate a result signal. It is by such computing that the desired beam characteristic is generated, between the acoustical input signals and the result signal. Often the at least two converters have omni-directional characteristics and it is only by the addressed computing that beamforming is achieved. Nevertheless, converters which have intrinsic beamforming ability may also be used but the desired transfer characteristic is conceived finally by the addressed computing.

[0007] Whenever a beam characteristic is realized by computing the electrical output signals of at least two acoustical to electrical converters or from more than two of such converters, whether a desired beam characteristic is accurately achieved depends from how accurately the involved converters provide for assumed predetermined transfer characteristics between their acoustical inputs and their electrical outputs.

Definition:

[0008]

- Two or more than two acoustical to electrical converters as microphones are considered to be matched if their real transfer characteristics between acoustical input signals and their electrical output signals is equal to such transfer characteristics as assumed when tailoring a desired beam characteristic.

- Two or more than two of such converters are considered to be substantially matched if due to adjustment of at least one of their electrical output signals it is achieved that their respective real transfer characteristics are less different from the assumed transfer characteristic than they are without such adjustment, i.e. given just by the intrinsic behavior of the converters.

- We understand under "matching" two or more than two acoustical to electrical converters, the process of mutually adjusting at least one characteristic feature of the transfer characteristic of at least one converter and so that the resulting real transfer characteristics of the at least two converters with the mutually adjusted electric output signals become less different from the assumed characteristics than they are without such adjustment. Characteristic features to be adjusted may e.g. be frequency response, thereby gain response and/or phase response. Thus, by the action of converter matching the converters become substantially matched, and not necessarily matched

[0009] Often, the desired beam characteristic is designed based on the assumption of identical transfer characteristics of the converters involved. Obviously, in such case the converters are made to be matched if the real transfer characteristics between acoustical input signals and respective possibly mutually adjusted electrical output signals are identical.

[0010] In this case too the process of matching the converters means mutually adjusting their electrical output signals so that the respective real transfer characteristics differ less than without such mutual adjusting and become, due to the mutual adjustment, in the ideal case, identical.

[0011] As a most common example - known as "delay and subtract" technique - beamforming is performed using at least two e.g. omni-directional converters which are mutually spaced by a predetermined distance, mutually delaying the output signals of the converters and subtracting the mutually delayed electrical signals which results in an overall beam characteristic which, with omni-directional converters, is of cardoid, hypercardoid, bidirectional or some other shape. Directivity of the resulting beam characteristic depends on one hand from the mutual distance of the converters, on the other hand from the possibly adjustable, thereby often automatically adjustable mutual delay, and from the accuracy with which the converters are matched.

[0012] If the two addressed converters are not matched the desired transfer characteristic will only be reached approximately.

[0013] Attempts have been made to match the at least two converters by mutual converter specimen selection or by mutually adjusting their electrical output signals, be it statically or dynamically, i.e. during operation of the beamformer.

[0014] Recently, dynamic matching is the preferred approach which allows accounting for time-varying transfer characteristics.

[0015] According to the DE-OS-19 822 021, which accords with the US-A 6 385 323, the electrical output signals of two microphone converters are fed via controlled matching amplifier units to a computing unit. The output signal of the computing unit has, with respect to acoustical input signals, a beam characteristic. The output signal powers resp. magnitudes of the matching amplifier units are averaged and the averaged signals compared by difference forming. The comparing result signal is fed to an analyzing and controller unit which controls the matching amplifier units. Thereby, the matching is performed in a negative feedback structure up to the comparing result of the two averaged signals vanishes. If this occurs the two input converters are considered to have been matched.

[0016] From the DE-OS-19 849 739 a similar approach as was discussed in context with the DE-OS 19 822 021 is known but in a feed-forwards structure. Significant characteristics as e.g. amplitude response or phase response of the analogue to digital converted output signals of two input converter microphones are compared and the output signal of one of the microphones is adjusted with respect to said characteristics as a function of the comparing result. It is further taught that whenever the two microphones have intrinsic beam characteristics directed in opposite directions, acoustical signals impinging laterally should lead to identical microphone output signals. Any deviation is then attributed to microphone mismatch and an appropriate adjustment is performed on the electric output signal of one of the microphones. Such ideal acoustical situation as only exploitable in free-field acoustical surrounding is apparently exploited for finding an appropriate optimum of pre-matching.

[0017] According to the WO 01/69968 the output signals of two microphones are computed. A result signal establishes with respect to the acoustical input signals a beamforming transfer characteristic. Each of the electrical output signals of the microphones is fed to a respective minimum estimation unit, the outputs thereof to a division unit. The result of the division controls a matching unit, namely a multiplying unit. It is recognized that because the microphones are often matched in free-field acoustical surrounding and not in-situ, the microphones can be mismatched when used in real life which degrades directionality. Matching is performed when the output signals of the microphones are minimal which is assigned to a "only noise" acoustical situation. This reference addresses multi-frequency band adaptive matching scheme.

[0018] From the EP 1 191 817 it is known to maintain a prevailing optimum directional transfer characteristic over time by forming a difference of averaged signals of the analog to digital converted microphone output signals and by feedback adjusting one of the digitalized microphone output signals to reduce the difference of the averaged signals.

[0019] In the US 6 272 229 mismatch of the microphone converters with respect to phase is also discussed. It is taught to provide acoustical delay compensation at two microphone output signals, thereby trying to compensate for time delays between acoustical signals impinging on the two microphones. A remaining time delay - after acoustical delay compensation - between the two output signals is assigned to microphone phase mismatch.

[0020] The US 2001/0038699 teaches to disable the directivity of the transfer characteristic, i.e. the beam characteristic of a two-microphone-based beamformer whenever "only noise" situation is recognized, thereby disabling one of the two microphones to reduce overall noise and maintaining only one microphone operative.

[0021] According to the DE-PS 19 918 883 which accords with the US 6 421 448 matching of two microphones is established with respect to frequency response by adjusting a filter arrangement between one of the microphone electrical outputs and a computing unit.

[0022] The present invention departs from the following recognitions:

[0023] Whenever a beamforming device or beamformer, which is based on at least two acoustical to electrical input converters, signals dependent on the output signals of these converters being computed, e.g. by delay-and-subtract operation, is applied in non-free field acoustical surrounding, such non-free field surrounding presents per se acoustical signal attenuation which varies as a function of spatial angle at which the acoustical source is seen from the acoustical input of the device. Such non-free field acoustical transfer characteristic, called "in-situ" characteristic, which varies with DOA is often important to be maintained as an informative entity. Generically, whenever according to known microphone matching approaches e.g. as described in the documents cited above, adjustment of the output signals of the converters is performed, this would lead - in the in-situ situation - to compensation of the in-situ transfer characteristic if fast time constants for the matching procedure were employed. Prior art literature like e.g. also US 6 385 323 or US 5 515 445 consider only aging, temperature, influence of dirt etc. as influencing factors for microphone matching though, i.e. they apply matching time constants in the range of minutes to days.

Definition

[0024] By matching time constant we understand the adaptation time constant to adapt the converters involved from one matching situation to another matching situation.

[0025] In hearing device appliances the head-related transfer function HRTF provides for an acoustic in-situ transfer characteristic between an acoustical source and the at least two converters, which differs from individual to individual and which varies significantly with varying DOA. If a sound source is thought to travel on a circular locus around an individual's head, the in-situ transfer characteristic between the acoustical source and individual's ear may vary by more than 10 dB as a function of DOA. The individual exploits such DOA dependency for localizing acoustical sources. Thus, such characteristic should not be spoiled by converter matching.

[0026] Prior art microphone matching algorithms employing long matching time constants to guard against aging, dirt influences etc. will not be able to provide sufficient dynamic matching in dependency of DOA without negatively influencing also HRTF related localization by the user of the hearing device.

[0027] It is one object of the present invention to provide for a matching technique for the at least two acoustical to electrical converters which maintains the effect of acoustical, surrounding-based transfer characteristics - in-situ transfer characteristics - to the converters.

[0028] This is achieved by the method for matching at least two acoustical to electrical converters, wherein signals respectively dependent on electrical output signals of the converters are computed to result in a result signal, the transfer characteristic between an acoustical signal impinging upon said at least two converters and said result signal being dependent on DOA of said acoustical signal upon the at least two converters. The method comprises matching the at least two converters for acoustical signals in dependency of an impinging direction of arrival within a range of direction of arrival upon said converters, said range being determined before performing said matching.

[0029] Thereby, the range of DOA of acoustical signals for which matching is performed is selected so that the in-situ transfer characteristic is known and in advance, as an example, is known to be neglectable. Techniques to evaluate the DOA of acoustical signals impinging on at least two acoustical to electrical converters of a beamforming device are known.

[0030] With respect to evaluation of the DOA we refer as an example to the WO 00/33634 which accords with the US patent application no. 10/180 585 of the same applicant as the present application. With respect to one possibility to monitor DOA the said WO 00/33634 as well as its US counterpart shall form by reference an integral part of the present application.

[0031] DOA evaluation is also strongly linked to time delay estimation for which numerous methods like cross-correlation, MUSIC, etc. are well known in the art. M. Brandstein "Microphone arrays", Springer, ISBN 3-540-41953-5 gives a nice overview over such methods. US 20010031053 shows another method for DOA estimation which is leaned on processes found in nature.

[0032] It has further been recognized that a range of DOA which is most suited to be exploited according to the present invention is where the desired transfer characteristic has minimum gain, i.e. around a "Null". This because signals impinging from the respective direction shall - according to the desired "Null" - be cancelled. Therefore, a realization form of the method according to the present invention, whereat the transfer characteristic has a minimum for a value of DOA, comprises matching the at least two converters for acoustical signals which impinge within the range determined before matching which includes such value of DOA.

[0033] Beamformers are further known which make use of at least two acoustical/electrical converters, signals dependent from their output signals being computed by a first computing and at least a second computing. The at least two computings result in respective first and second result signals. Thereby, a first transfer characteristic between an acoustical signal impinging on the at least two converters and the first result signal and which is dependent on DOA is differently dependent on DOA than a second transfer characteristic between the acoustical input signal and the second result signal. Such beamforming devices are e.g. realized by the so-called Griffith Jim-based beamformers as exemplified e.g. in the US 5 473 701 to AT&T.

[0034] According to an embodiment of the present invention in such a case matching is performed independently for the addressed first and at least one second computing, for acoustical signals which respectively impinge from ranges of DOA determined before matching upon the at least two converters. These ranges may be selected to be equal or to be different.

[0035] In an embodiment of the present invention matching is performed selectively in frequency bands determined before matching, whereby in a further embodiment of the invention analog to digital and time-domain to frequency-domain conversion is performed between the electrical output of the at least two converters and computing.

[0036] Attention is drawn to the enclosed **Attachment A** which is a yet unpublished European patent application with application No. 04 006 073.3 filed March 15, 2004 and which accords to a US application filed same date with a yet unknown Serial Number. This unpublished and therefore annexed patent application is to be considered as a part of the present description by reference with respect to the following subject matter:

[0037] In the Attachment A a method for suppressing feedback between an acoustical output of an electrical/acoustical output converter arrangement and an acoustical input of an acoustical/electrical input converter arrangement of a hearing device is addressed. Thereby, acoustical signals impinging on an input converter arrangement are converted into a first electrical signal by a controllably variable transfer characteristic which is dependent on the angle (DOA) at which the acoustical signals impinge on the input converter arrangement. The first electrical signal is then processed and a signal resulting from such processing is applied to the output converter.

[0038] Thus, and with an eye on the present description the following may be established:

[0039] The acoustical/electrical input converter arrangement as addressed in the **Attachment A**, wherein acoustical signals impinging on the input converter arrangement are converted into a first electrical signal by a controllably variable transfer characteristic which is dependent on the angle at which the acoustical signals impinge on the input converter arrangement, accords in the present description to the at least two acoustical to electrical converters, computing and generating the result signal.

[0040] When applying the device according to the present invention to hearing devices as addressed above, the result signal is operationally connected via a processing unit to an electrical/acoustical output converter arrangement. Further, the teaching according to the **Attachment A** addresses a method for suppressing feedback between the output of such electrical/acoustical output converter arrangement and the input of the at least two converters as addressed in the present description.

[0041] According to the present application as was already addressed the at least two input converters are to be matched during operation, i.e. automatically, whereby in fact the real transfer characteristic is adjusted. This accords with the definition in **Attachment A** of an adaptive beamformer unit.

[0042] If according to one embodiment of the present invention the result signal is operationally connected to an output electrical/acoustical converter as of a hearing device and there is provided, as described in the **Attachment A** in details, a feedback compensator, the input of which being operationally connected to the input of the output converter arrangement, the output of which being fed back, the complex task of estimating the feedback signal to be suppressed by the feedback compensator e.g. by correlation leads to the fact that the feedback compensation process has a relatively long adaptation time constant to adapt from one feedback situation to be suppressed to another by appropriately varying the loop gain of the feedback loop. As described in the **Attachment A** such an adaptation time constant is customarily in the range of hundreds of msec.

[0043] The matching process which is addressed in the present application defines as well for an adaptation time constant of the adaptive beamformer. The adaptation time constant for "matching adaptation" is significantly shorter than the adaptation time constant as realized by the feedback compensator. Therefore, and if according to one aspect of the present invention a feedback compensator is provided as explained in detail in the addressed **Attachment A**, the same problems arise as also explained in the addressed **Attachment A**, namely the problem that the feedback compensator may not follow quick changes of feedback situations which are caused by the short adaptation time constants of matching adaptation. Thus, and according to one aspect of the present invention, this is resolved by that embodiment of the present invention, wherein the addressed result signal is operationally connected to an electric input of an electrical to acoustical converter and which comprises feeding back an electric feedback compensating signal which is dependent on an input signal to the electrical to acoustical converter and superimposing the fed-back signal to the result signal, wherein further the adaptation rate of matching according to the present invention is controlled in dependency of the loop gain along the feedback signal path.

[0044] The skilled artisan will recognize also from the **Attachment A** or the respective applications once published, how to realize the just addressed embodiment of the invention.

[0045] As was addressed above prior art matching is accomplished with matching time constants τ which are very long, namely in the range of minutes up to days. Thereby, such matching may not cope with converter matching needs which arise at short term.

[0046] This is remedied by the present invention under a second aspect by providing for a method for matching at least two acoustical to electrical converters, signals dependent on the electrical output signals of the converters being computed to result in a result signal and wherein the transfer characteristic between an acoustical signal impinging upon the at least two converters and the result signal is dependent on direction of arrival of the acoustical signal on the at least two converters, wherein matching of the converters is performed with a matching time constant τ, for which there is valid:

$$0 < \tau \leq 5 \text{ sec.}$$

[0047] Thereby, in a further embodiment there is established

$$0 < \tau \leq 1 \text{ sec.}$$

[0048] And in a still further embodiment

$$0 < \tau \leq 100 \text{ msec.}$$

[0049] A beamforming device according to the present invention comprises at least two acoustical to electrical converters and at least one computing unit, the electrical output of the converters being operationally connected via a matching unit to inputs of the at least one computing unit. Thereby, the output of the beamforming device is operationally connected to the output of the at least one computing unit. The computing unit further generates a signal which is indicative of DOA of an acoustical signal which impinges on the at least two converters. The device further comprises a matching control unit which generates a matching control signal which is operationally connected to a control input of the matching unit. The signal which is indicative of DOA is further operationally connected to a control input of the matching control unit, which further has at least two inputs which are operationally connected to respective outputs of the at least two converters, in feedback structure downstream the matching unit, in feed-forwards structure upstream the matching unit.

[0050] Under a second aspect of the present invention there is provided a beamforming device comprising at least two acoustical to electrical converters and at least one computing unit, the electrical output of said converter being operationally connected via a matching unit to inputs of said at least one computing unit, the output of said beamforming device being operationally connected to the output of said at least one computing unit, a matching control unit generating a matching control signal operationally connected to a control input of the matching unit, said matching unit comprising at least two inputs operationally connected to the outputs of said at least two converters upstream or downstream said matching unit and wherein said matching control unit generates the matching control signal so as to match the at least two converters with a matching time constant τ for which there is valid:

$$0 < \tau \leq 5 \text{ sec.}$$

[0051] In a further embodiment under this second aspect the matching time constant τ is:

$$0 < \tau \leq 1 \text{ sec.}$$

[0052] In a still further embodiment there is valid:

$$0 < \tau \leq 100 \text{ msec.}$$

[0053] It is further to be noted that when we speak of a value or of a frequency band which is determined before matching is performed, the meaning of "before" encompasses a long time span before, e.g. when a respective device is fitted or even is manufactured up to a very short time span when such a value or frequency band is determined dynamically in situ just before the respective matching is performed.

[0054] Preferred embodiments of the present invention shall now be exemplified with the help of figures. These as well as the appending claims will also reveal to the skilled artisan additional embodiments of the device according to the invention.

[0055] The figures show:

Fig. 1 schematically and simplified, by means of a signal-flow/functional block diagram, an embodiment of the device according to the present invention performing the method according to the invention;

Fig. 2 a schematic representation of steps as performed by the method and device according to fig. 1;

Fig. 3 in a representation in analogy to that of fig. 1, the implementation of the device of fig. 1, e.g. in a hearing device as an embodiment of the invention with feedback compensation, and

Fig. 4 a further embodiment of a device according to the present invention operating according to the method of the present invention, again in a representation in analogy to that of fig. 1.

[0056] According to fig. 1 a number of acoustical to electrical converters, as shown two such converters 1a and 1b, have electrical outputs A_{1a}, A_{1b} which are operationally connected to inputs E_{3a} and E_{3b} of a matching unit 3. As shown in dashed lines within matching unit 3 signals which are applied to the inputs E_{3a} and E_{3b} are adjusted with respect to at least one of their characteristics, e.g. with respect to frequency response, amplitude and/or phase response or other characteristic features.

[0057] Respective adjusting members are provided in unit 3, e.g. as shown in channel a or b or in both channels a and b. The outputs A_{3a} and A_{3b} are operationally connected to inputs E_{7a} and E_{7b} of a computing unit 7 which has an output A_7 and an output A_{DOA}.

[0058] Within computing unit 7 on one hand and as schematically shown by unit 7_{BF} beamforming is computed from the signals applied to the inputs E_{7a}, E_{7b} e.g. by delay-and-subtract computing. The result of beamforming is fed to output A_7 as a result signal of the beamforming operation.

[0059] Additionally, in computing unit 7 the direction of arrival DOA of acoustical signals impinging upon the converters 1a and 1b is computed from the signals applied to E_{7a}, E_{7b} resulting in an output signal fed to output A_{DOA} of computing unit 7 which is indicative of DOA of the addressed acoustical signals. In unit 7_{DOA} performing monitoring of the DOA is e.g. realized as described in the WO 00/33634 which was already mentioned above or as taught by the following publications:

M. Brandstein "Microphone arrays", Springer, ISBN 3-540-41953 or US 2001003053.

[0060] At the output A_{DOA} of computing unit 7 there is generated a signal which is indicative of the direction of arrival DOA. This signal is operationally connected to a comparator unit 9, where it is checked, whether the instantaneously evaluated DOA signal is within a range $\pm\Delta$DOA around a value DOA_S. Determination, whether the actual DOA signal is within this range $DOA_S \pm \Delta DOA$ is performed by comparing the DOA indicative signal from the output A_{DOA} with a signal range which is preset at input E_{9C} of unit 9. Whenever it is detected in unit 9 that the prevailing DOA signal is within the predetermined range, unit 9 generates at an output A_9 a control signal which is operationally connected to a control input E_{11C} of a matching control unit 11. The matching control unit 11 has two further inputs E_{11a} and E_{11b} which are operationally connected to the electric output A_{1a} and A_{1b} of the respective converters. The signals applied to the input E_{11a} and E_{11b} are compared as shown in block 11 e.g. by difference forming and an output signal is generated at output A_{11} of matching control unit 11, which is dependent on the result of such comparison. As further schematically shown within unit 11 the signal applied to control input E_{11c} enables the comparison result dependent signal to become effective via output A_{11} on adjustment control input E_{3c} of matching unit 3, controlling the adjustant members provided in matching unit 3. Thereby, as a function of the comparing result in matching control unit 11, the at least two signals which are fed to the computing unit 7 at E_{7a} and E_{7b} are adjusted to become less different.

[0061] Whereas fig. 1 shows a feed forwards structure the same technique may be realized in a feed-back structure (not shown) by connecting the inputs E_{11a} and E_{11b} not to the outputs of the converters 1a and 1b upstream unit 3, but instead to the outputs A_{3a} and A_{3b} downstream matching unit 3.

[0062] In fig. 2 processing as performed with the device and method exemplified with the help of fig. 1 shall further be explained. Representation (a) shows as an example the transfer characteristic in polar representation of an omnidirectional converter as of converter 1a of fig. 1. Representation (b) shows such transfer characteristic again as an example of the second converter as of 1b of fig. 1. Based on these two converter-intrinsic omnidirectional transfer characteristics, beamforming within computing unit 7 leads e.g. to the cardoid transfer characteristic as shown in representation (c) which is e.g. realized by the delay-and-subtract method.

[0063] Within computing unit 7 and as shown in fig. 1 by block 7_{DOA}, the instantaneously prevailing DOA is estimated as shown in representation (d) to be α. The range, which is determined before performing matching, $DOA_S \pm \Delta DOA$ as also shown in representation (d) is exemplified with $DOA_S = 0$, at which a "Null" of the desired transfer characteristic as of representation (c) is expected. Only then when the estimated DOA according to α is within the range $DOA_S \pm \Delta DOA$, with an eye on fig. 1, matching of the converters is initiated by means of the signal generated at the input E_{11C}. Techniques which are applicable for mutually adjusting the signals in matching unit 3 are well-known as has been shown by the referenced publications in the introductory part of the present description. Accordingly, the matching control unit 11 is realized to provide for the desired dependency between the comparison result of comparing the signals applied to the inputs E_{11a} and E_{11b} and adjustment of the respective adjusting members in unit 3.

[0064] Instead of enabling/disabling, practically in a hard switching manner, matching of the converters via matching control unit 11 it is possible to softly weigh the effect of the comparing result computed in matching control unit 11 upon the adjusting members in matching 3 e.g. as a function of deviation between estimated DOA and DOA_S as determined before performing matching. Such weighing may e.g. be realized so that such effect becomes the weaker resp. the matching frozen the more that the estimated DOA deviates from DOA_S.

[0065] In fig. 3 a further embodiment of a device according to the present invention operating according to the method of the invention is shown. The same reference numbers are used in fig. 3 as in fig. 1 for elements which have already been described in context with fig. 1.

[0066] The unit comprising the converters 1a, 1b, matching unit 3, computing unit 7, matching control unit 11, provides for an adaptive beamformer unit 20_A, whereby being adapted by adjusting the overall transfer function by converter matching.

[0067] The output A_7 of the adaptive beamformer 20_A is operationally connected to a superimposing unit 20_{AP}.

[0068] Attention is drawn to the convention with respect to the reference numbers applied in fig. 3. The same reference numbers are used as used in the **Attachment A**, fig. 3, which latter teaches in more details the technique as also applied in the embodiment of fig. 3 of the present invention. Nevertheless, these linking reference numbers are indexed with "AP" (for Appendix).

[0069] The output of the superimposing unit 20_{AP} is input to processing unit 14_{AP}, the output thereof being operationally connected to the input of an electrical to acoustical converter arrangement 16_{AP}. Thereby, the combined structure of beamformer 20A, processing unit 14_{AP} and electrical to acoustical converter arrangement 16_{AP} is a structure typical e.g. in hearing device applications.

[0070] A compensator unit 18_{AP} has an input operationally connected to the input of the converter arrangement 16_{AP} and an output operationally connected to one input of the superimposing unit 20_{AP}. The negative feedback loop with compensator unit 18_{AP} provides for compensation of acoustical feedback from the acoustical output of converter arrangement 16_{AP} to the acoustical input of the converters 1a, 1b.

[0071] As schematically shown in fig. 3 the compensator unit 18_{AP} has an output A_{GAP}, whereat a signal is generated which is indicative of the loop gain of the negative feedback loop. This loop gain may e.g. be estimated by multiplying the linear gains along the loop which primarily consists of the compensator unit 18 and of processing unit 14_{AP} or by adding these gains in dB.

[0072] The loop gain indicative signal at output A_{GAP} is fed to a control input C_{12RAP} of the adaptive beamformer 20_A and therein to a control input of matching control unit 11. By means of the loop gain indicative signal applied to this control input, the matching adaptation rate at matching unit 3 and via matching control unit 11 is slowed down at least down to the adaptation rate of compensator unit 18_{AP} in dependency of the prevailing feedback effect and thus of the loop gain of compensator unit 18_{AP}. Thereby, combination of the beamformer unit 20A with automatically matched converters 1a and 1b according to the present invention and of feedback compensation becomes feasible.

[0073] In fig. 4 a further embodiment of the present invention is shown. Again, reference numbers which were already used in context with fig. 1 or 3 are used for elements which have already been described. According to the embodiment of fig. 4 the outputs A_{1a} and A_{1b} of the at least two converters 1a and 1b are operationally connected to a first matching unit 3_I and to a second matching unit 3_{II}.

[0074] The outputs of the two matching units 3_I and 3_{II} are operationally connected to respective computing units 7_I and 7_{II}. At the output A_{7I} there appears a first result signal. Between an acoustical input signal impinging on the converters 1a and 1b and the first result signal at output A_{7I} there prevails a first transfer characteristic which is differently dependent on DOA than a second transfer characteristic which prevails between the acoustical input signal upon converters 1a and 1b and a signal generated at output A_{7II} of the second computing unit 7_{II}.

[0075] Thus, in fact based on the converters 1a and 1b two beamformers are realized with different beam characteristics. Matching is performed independently at both beamformers as follows:

[0076] Matching of the converters with respect to first beamformer I is performed via unit 9_I, matching control unit 11_I in analogy to the one beamformer technique of fig. 1. Further in complete analogy matching of the converters 1a and 1b with respect to the second beamformer II is performed via unit 9_{II}, matching control unit 11_{II}. As may be seen in fig. 4 in opposition to the representation in fig. 1 a feedback structure is shown in that the outputs of the respective matching units 3_I and 3_{II} are fed for comparison purposes to the matching control units 11_I and 11_{II}.

[0077] In all the embodiments of the invention signal processing may be performed in analog or digital or hybrid technique. Converter matching selectively in frequency bands which are determined before performing matching is simplified by signal processing in the frequency domain.

[0078] Due to the fact that according to the one aspect of the present invention converter matching is only then performed when an acoustical signal impinges on the input converters within a range of DOA and this range may be selected in an optimum direction with an eye on in-situ situation, it is achieved that automatic in-situ converter matching is feasible without affecting the effects of the in-situ acoustic situation.

[0079] As was already addressed above generically matching time constants for direction of arrival controlled matching as was described with the help of figures 1 to 4 may be performed with a matching time constant τ for which there is valid:

$$0 < \tau \leq 5 \text{ sec.}$$

[0080] Thereby, such time constant τ may be even selected to be:

$$0 < \tau \leq 1 \text{ sec.}$$

or even to be

$$0 < \tau \leq 100 \text{ msec.}$$

[0081] Nevertheless and irrespectively of controlling converter matching in dependency of direction of arrival, more generically, a beamformer technique is addressed under a second aspect which makes use of at least two acoustical to electrical converters and where converter matching is performed with matching time constants τ for which the addressed ranges are valid.

Attachment A

Feedback suppression

[0082] The present invention deals with a method for suppressing feedback between an acoustical output of an electrical/acoustical output converter arrangement and an acoustical input of a acoustical/electrical input converter arrangement of a hearing device, wherein acoustical signals impinging on the input converter arrangement are converted into a first electrical signal, by a controllably variable transfer characteristic and which is dependent on the angle at which said acoustical signals impinge on the input converter arrangement. The first electrical signal is processed and a resulting signal is applied to the output converter. There is further provided an electrical feedback-compensating signal, generated in dependency of the result signal which is applied via a feedback signal path upstream the processing.

Definition

[0083] A unit to which the output of the input converter arrangement is input and which provides a signal transfer characteristic to its output which has an amplification dependent on spatial angle at which acoustical signals impinge on the acoustic input of the input converter arrangement is called a beamformer unit. The transfer characteristic in polar representation is called the beam.

[0084] An adaptive beamformer unit is a beamformer unit, the beam generated therefrom being controllably variable.

[0085] From the EP 0 656 737 there is known such a method which nevertheless does not apply beamforming. The input of a feedback-compensator is operationally connected to the input of the output converter arrangement of the device, the output of the compensator is operationally connected to the output of the input converter arrangement, thereby forming a feedback signal path.

[0086] Due to the complex task of estimating the feedback-signal to be suppressed e.g. by correlation at the feedback-compensator, the feedback-compensation process has a relatively long adaptation time constant to adapt from one feedback situation to be suppressed to another by appropriately varying its gain. Such an adaptation time constant is customarily in the range of hundreds of milliseconds.

[0087] Feedback signals to be suppressed impinge upon the input acoustical/electrical converter arrangement substantially from distinct spatial angles. As schematically shown in Fig. 1, a behind-the-ear hearing device 3 with an input converter arrangement 5 applied at the pinna 1 of an individual, experiences feedback to be suppressed from a distinct direction as shown at d1. An in-the-ear hearing device 7 according to Fig. 2 which has, as an example, a vent 9 and two acoustical ports 11 to the input converter arrangement, experiences feedback signals to be suppressed from the distinct directions d2.

[0088] Therefore, a further approach for suppressing feedback is to install high signal attenuation between the input and the output converter of the device for signals which impinge on the input converter under such distinct spatial angles. This accords with applying a beamformer technique generating a beam having zero or minimum amplification at such angles.

[0089] Hearing devices which have adaptive beamformer ability are known e.g. from the WO 00/33634. For feedback suppression at a hearing device with adaptive beamforming ability, it seems, at first, quite straight forward to combine on the one hand feedback compensation techniques as e.g. known from the EP 0 656 737 with adaptive beamformer technique as e.g. known from the WO 00/33634 and thereby to place minimum amplification of the beam at those angles which are specific for feedback signals to be suppressed impinging on the input converter. This especially because these angles are clearly different from the target direction range within which maximum amplification of the beam is to be variably set.

[0090] Thereby, it has to be noted that the adaptation time constant of an adaptive beamformer unit is considerably smaller, in the range of single to few dozen milliseconds, than the adaption time constant of a feedback-compensator which is, as mentioned above, in the range of hundreds of milliseconds.

[0091] One approach is known where a beamformer unit is provided, the input thereof being operationally connected to two mutually distant microphones of an input converter arrangement. As both spaced apart microphones experience the feedback signal to be suppressed differently, two feedback compensators are provided with inputs operationally connected to the input of the output converter arrangement. The respective output signals are superimposed to the respective output signals of the two microphones.

[0092] The fact that the adaptation time constant of the beamformer unit is much shorter than the adaptation time constant of the compensators does not pose a problem in this configuration, because the fast adapting beamformer unit is placed within the closed feedback loop formed by the feedback-compensation feedback paths.

[0093] Nevertheless, this known approach has the serious drawback that for each of the microphones one compensator feedback path must be provided which unacceptably raises computational load.

[0094] A further approach for beamformer/feedback-compensation combination is known from M. Brandenstein et al. "Microphone arrays", Springer Verlag 2001. Here the feedback compensation path is fed back to the output of the beamformer unit. By this approach only one compensation path is necessary and thus computational load is reduced. Nevertheless, here the fast adapting beamformer is outside the negative feedback loop. Thus, whenever the adaptive beamformer is controlled to rapidly change its beam pattern, the compensator will not be able to adequately rapidly deal with the new situation of feedback to be suppressed.

[0095] Therefore, M. Brandenstein et al. "Microphone arrays" considers this approach as, at least, very difficult to realise.

[0096] A third approach is proposed in M. Brandenstein et al. as mentioned and in W. Herbold et al. "Computationally efficient frequency domain combination of acoustic echo cancellation and robust adaptive beamforming". A generalised side lobe cancelling technique for the beamformer is used whereat only a not-adaptive beamformer is placed upstream the compensation feedback path, thus eliminating the adaptation time problem as well as double computational load. Nevertheless, by this approach placing minimum amplification of the beam in the direction of feedback signal arrival may not be realised.

[0097] It is an object of the present invention to provide a method for suppressing feedback as addressed above at a hearing device which has an adaptive beamformer on the one hand, and a feedback compensator on the other hand, thereby avoiding the drawbacks as addressed above.

[0098] This is achieved on the one hand by superimposing the fed back feedback compensating signal to the signal downstream the beamforming, and, on the other hand, by controlling the adaptation rate of beamforming in dependency of the gain along feedback signal path with the compensator.

[0099] Thus, there is proposed a method for suppressing feedback between an acoustical output of an electrical/acoustical output converter arrangement and an acoustical input of an acoustical/electrical input converter arrangement of a hearing device, wherein acoustical signals impinging on the input converter arrangement are converted into a first electric signal by a controllably variable transfer characteristic which is dependent on the angle at which said acoustical signals impinge on said input converter arrangement. The first electric signal is processed and a resulting signal is applied to the output converter arrangement. The feedback to be suppressed is compensated by a feedback compensating signal which is generated in dependency of the resulting signal and is fed back by a feedback signal path to a location along the signal path upstream the processing. Thereby, the feedback-compensating signal is fed back to the first electric signal - thus downstream the beamformer - and the adaptation rate of converting to variations of the transfer characteristic - and thus of beamforming - is controlled in dependency of gain along the compensator feedback signal path.

Definition

[0100] We understand by the adaptation rate of the adaptive beamformer unit the speed with which the beamformer unit reacts on an adaptation command to change beamforming operation as e.g. changing target enhancement or noise suppression direction. The adaptation rate accords with an adaptation time constant to change from one beamforming polar pattern to another.

[0101] We understand by the adaptation rate of feedback-compensating the rate with which the respective compensator reacts on a detected change of feedback situation until the compensator has settled to a new setting. The compensator thereby estimates the prevailing situation of feedback to be suppressed e.g. by a correlation technique between the signal applied to the output converter arrangement and the signal received from the input converter arrangement as e.g. described in the EP 0 656 737. The adaptation rate of the compensator accords with an adaptation time constant too. Whenever the loop gain along the compensating feedback signal path increases, this is caused by an increasing amount of feedback to be suppressed and thus to be compensated. This means that the adaptation rate of the beamformer unit is to be slowed down so that the compensator feedback signal may model the response of the beamformer unit too. Thus, in a preferred embodiment, the adaptation rate of converting i.e. of beamforming is slowed down with increasing loop gain along the feedback signal path.

[0102] As was addressed above, feedback signals, which are acoustical and which have to be suppressed, impinge on the acoustical input of the input converter arrangement substantially and dependent on the specific device at specific angles. Thus, in a most preferred embodiment of the method according to the present invention, amplification of the transfer characteristic representing beamforming is minimized at one or more than one specific angles which accord to angles at which the feedback to be suppressed predominantly impinges on the input converter arrangement.

[0103] Thus, and considered in combination with slowing down the adaptation rate of beamforming with increasing gain along feedback compensation fed back signal path, it becomes apparent that the compensator may still model the beamformer without losing the established minimum or minima in the direction of the said specific angles.

[0104] Further, it has to be noted that the feedback to be suppressed is a narrow band acoustical signal, thus in a further improvement of the method according to the present invention, it is not necessary - so as to deal with a feedback to be suppressed - to control and especially to slow down the adaptation rate of beamforming conversion in the entire frequency range beamforming is effective at, but it suffices to controllably adapt the adaptation rate of the beamforming conversion at frequencies which are significant for the feedback signal to be suppressed. Therefore, in a further preferred embodiment of the present invention, controlling of the adaptation rate of the beamforming conversion is performed frequency selectively.

[0105] In spite of the fact that the principal according to the present invention may be applied at hearing devices where signal processing is performed in analog technique, it is preferred to perform the method in devices where signal processing is performed digitally. Thereby, and in view of the addressed preferred frequency selective control, in a most preferred embodiment, at least signal processing in the beamforming conversion as well as along the feedback compensation path, is performed in frequency domain, whereby time domain to frequency domain conversion may be realised in a known manner, be it by FFT, DCT, wavelet transform or other suitable transforms. The respective reconversion for the signal applied to the output converter arrangement is performed with the respective inverse processes. The adaptation rate is controlled at selected frequencies in dependency of the compensator gain at these selected frequencies. Thereby the following approach is achieved:

[0106] As beamforming is only effective with respect to the feedback to be suppressed at specific frequencies or at a specific frequency band on the one hand the control of the adaptation rate of beamforming is in fact only to be performed at these specific frequencies or for the addressed frequency band. Further, selecting minimum amplification at the specific feedback impingement angles must be provided at the beamformer only for the specific frequencies or for the frequency band of the feedback to be suppressed too. Thus, this leads to the recognition that in fact beamforming may be subdivided in beamforming for frequencies which are not significant for the feedback to be suppressed and beamforming for frequencies or the frequency band which is specific for the feedback signal to be suppressed. Thus, beamforming in the addressed specific frequencies may be performed and its adaptation rate controlled independently from tailoring beamforming at frequencies which are not specific for the feedback signal to be suppressed. This beamforming may be performed at adaption rates which are independent from feedback compensation and thus faster and which generates a beam which is not dealing with the specific impinging angles of the feedback signal to be suppressed.

[0107] Therefore, in a further preferred embodiment of the method according to the present invention, performing controlling of beamforming is done selectively at frequencies which are significant for the feedback to be suppressed. Further preferred minimalising the amplification of the beamforming transfer characteristic is only done at specific angles in a frequency selection manner. In fact two independent beamforming actions are superimposed, a first dealing with the generically desired beamforming behaviour, a second dealing with feedback suppression as concerns frequencies and as concerns beamshaping. It becomes possible e.g. to switch off first beamforming, thereby maintaining the second and thereby preventing acoustical feedback to become effective. The method according to the present invention may be applied to behind-the-ear hearing devices or to in-the-ear hearing devices, monaural or binaural systems, and further may be applied to such devices which are conceived as ear protection devices i.e. protecting the human ear from excess acoustical load, or to hearing improvement devices be it just to improve or facilitate hearing by an individual, or in the sense of a hearing aid, to improve hearing of a hearing impaired individual.

[0108] It is to be noted that feedback caused not by acoustical but by electrical or mechanical reasons is often fed into the microphones of the input converter arrangement with equal gains and phases, thus appearing to originate from a direction perpendicular to the port axis of the input converter arrangement. In an endfire array, as typically used in hearing instruments, this conforms to a 90° direction or arrival, and may be suppressed by a beamformer arrangement according to the present invention as well.

[0109] To resolve the object as mentioned above, there is further, and according to the present invention, provided a hearing device which comprises:

- an acoustical/electrical input converter arrangement and a adaptive beamformer unit generating at an output an electric output signal dependent on acoustical signals impinging on said input converter arrangement and in dependency of angle at which said acoustical signals impinge, said beamformer unit having a first control input for varying beamforming characteristics and a second control input for controllably adjusting adaptation rate;
- a processing unit with an input operationally connected to the output of said beamformer unit with an output operationally connected to an input of an electrical/acoustical output converter arrangement;
- a feedback compensator unit, the input thereof being operationally connected to said input of said electrical/acoustical output converter arrangement, the output thereof being operationally connected to the input of said processing unit and having a loop gain output, said loop gain output being operationally connected to said second control input of said beamformer unit.

[0110] Preferred embodiments of the method according to the present invention, as well as of a hearing device according to the present invention, shall additionally become apparent from the following detailed description of preferred embodiments with the help of further figures and from the claims. The figures show:

Figs. 1 & 2: as discussed above, schematically specific angles at which feedback signals impinge on the acoustical input port of outside-the-ear (Fig. 1) and in-the-ear (Fig. 2) hearing devices.

Fig. 3: by means of a simplified functional block/signal flow-diagram, a device according to the present invention operated according to the method of the present invention.

Fig. 4: in polar diagram representation preferred beamforming at the device according to Fig. 3 taking into account specific angles with which the feedback to be suppressed impinges on the acoustic input as exemplified in the Figs. 1 or 2.

Fig. 5a: as an example and quantitatively, beamforming by the device of Fig. 3 at specific frequencies which are significantly present in the feedback signal to be suppressed.

Fig. 5b: beamforming at the device of Fig. 3 for frequencies which are not significantly present in the feedback signal to be suppressed.

[0111] In Fig. 3 there is schematically shown, by means of a signal flow-/functional block-diagram a device according to the present invention, whereat the method according to the invention is realised. The device comprises an input acoustical/electrical converter arrangement 10, which cooperate with a beamformer unit 12. The conversion characteristics of the input converter arrangement 10 together with signal processing in beamformer unit 12 provides a beamformer characteristic between acoustical input E_{10} to input converter arrangement 10 and electrical output A_{12} of the beamformer unit 12. The beamformer unit 12 has an adaptation control input C_{12A} and α adaptation rate control input C_{12R}.

[0112] The transfer characteristic between E_{10} and A_{12} has an amplification which is dependent on the angle α at which acoustical signals impinge on the acoustical port of input converter 10. Thus, there is generated by the combined units 10 and 12 a beam characteristic as exemplified with B in unit 12.

[0113] As further schematically shown by the variation arrow V within block 12, the transfer characteristic, in polar representation the beam B, may be varied with respect to its characteristics as e.g. with respect to target direction, maximum amplification etc. as shown in dotted line within block 12. Variation of the beam characteristic B is controlled by control input C_{12A} which latter is, as shown in dotted line, normally connected to a processing unit 14 for adapting the beam characteristic B e.g. to prevailing acoustical situations automatically or program controlled or by an individual wearing the hearing device.

[0114] Beamforming units which may be adapted are known. One example thereof is described in the WO 00/33634.

[0115] Variation of the beam characteristic B may also be caused at the beamformer itself, i.e. by beamformer internal reasons.

[0116] Therefore, it must be emphasised that the input C_{12A} and control signals applied thereto are merely a schematic representation of beam characteristic variation ability or occurrence.

[0117] The electrical output of beamforming unit 12, A_{12}, is operationally connected to an input E_{14}, of the signal processing 14 unit whereat input signals are processed and output at an output A_{14} operationally connected to an electric input E_{16} of an output electrical to acoustical converter arrangement 16 so as to provide desired ear protections or hearing improvement to the individual carrying such device. We understand under ear protecting ability the ability of reducing or even cancelling acoustical signals which impinge on the input converter arrangement 10, so as to protect individual's hearing or even provide the individual with silent perception in non-vanishing acoustical surroundings. Under hearing improvement, we understand the improvement of individual's hearing in an acoustical surrounding, be it for customary applications of normal hearing individual or be it in the sense of hearing aid to improve individual's impaired hearing.

[0118] As perfectly known to the skilled artisan, one ongoing problem in context with such hearing devices is the acoustical feedback AFB between the acoustical output of the output converter 16 and acoustical input E_{10} of the input converter arrangement 10. As principally known e.g. from the EP 0 656 737, there is provided a feedback compensator 18 whereat the prevailed acoustical feedback AFB, which is to be suppressed, is estimated e.g. with a correlation technique, correlating the signal applied to output converter 16 with a signal dependent on the output of input converter 10 as shown in dashed line at A. Thereby the gain G of compensator 18 is estimated so a to compensate for the AFB by negative feedback.

[0119] By means of compensator unit 18, a signal as predicted is fed back to the input of processor unit 14 downstream the output of beamformer unit 12 so as to compensate for the feedback AFB. As shown in Fig. 3, the compensator unit 18 has an input E18 which is operationally connected to the output A_{14} of the processing unit 14 and has an output A_{18} which is superimposed to the output E_{12} of beamformer unit 12, the result of such superimposing being input to input E_{14} of processing unit 14.

[0120] Customarily, the compensator unit 18, which computes estimation of the acoustical feedback to be suppressed, has an adaptation rate in the range of several hundred ms and is thus considerably slower than the adaptation rate of beamforer unit 12. Thus without additional measures according to the present invention, whenever the beamformer unit 12 is controlled or caused to vary its beamforming characteristic B as schematically represented by a control at input C_{12A}, the compensator 18 will not be able to accurately rapidly deal with the varied situation with respect to acoustical feedback AFB.

[0121] Therefore, there is provided a control of the adaptation rate of beamformer unit 12 which control is performed by the compensator unit 18, according to Fig. 3 at control input C_{12R}. Whenever the feedback signal loop gain via compensator 18 rises, indicating the increase in acoustical feedback AFB to be suppressed, the adaptation rate or time constant of beamformer unit 12 is lowered to or below the adaptation rate of compensator unit 18.

[0122] The loop gain may at be least estimated e.g. by multiplying the linear gains along the loop, primarily consisting of the compensator 18 and the processing unit 14 in Fig. 3 or by adding these gains in dB.

[0123] Thereby, it is prevented that an adjustment of the beamformer unit 12 with respect to its beamforming characteristic B may not be dealt with by compensator unit 18.

[0124] Thus, in fact, adaptation rate control of beamformer unit 12 is performed in dependency of the loop gain along the feedback loop with compensator unit 18. The rate control input C_{12R} to beamforming unit 12 is operationally connected to a loop gain output A_G of unit 18. With the embodiment according to the present invention as shown in Fig. 3, it becomes possible to slow down the adaptation rate of the beamformer unit 12 at least down to the adaptation rate of the feedback compensator unit 18 in dependency of prevailing feedback of compensator 18.

[0125] Thereby, combination of adaptive beamforming and feedback compensating becomes feasible.

[0126] As has already been mentioned, the direction with which acoustical feedback signals AFB to be suppressed impinge on the acoustical port of the input converter 10 is specific. Therefore, at the beamformer unit 12, there is generated a beam characteristic B_{AFB}, as shown in Fig. 4, which has minimum amplification for these specific angle or, as shown e.g. for an in-the-ear hearing device, at two specific angles α_{AFB}. Thus and in addition to compensation of AFB by compensator unit 18, beamforming is realised with minimum amplification for those spatial angles α_{AFB} with which the acoustical feedback AFB to be suppressed impinges on the input converter 10.

[0127] Further, it has to be noticed that acoustical feedback AFB to be suppressed occurs substantially within a specific frequency band. This frequency band is dependent, among others, on the specific output converter 16 used, the type of device e.g. in-the-ear or outside-the-ear device. Therefore, in a further improved embodiment, overall feedback suppression may be performed within that specific frequency band, thereby leaving beamforming in frequencies not within this specific frequency band unaffected and tailored according to needs different from acoustic feedback suppression. According to Fig. 5 (a), beamforming $B_{\overline{AFB}}$ for minimum amplification of acoustical feedback AFB to be suppressed, is performed frequency selectively for frequencies $f_{\overline{AFB}}$ of the acoustical feedback signal AFB.

[0128] Beamforming for frequencies f_{AFB} which are not significantly present in the acoustical feedback AFB is performed by a second beamforming $B_{\overline{AFB}}$ which may be selected independently from B_{AFB}.

[0129] In fact, two independent beam forms are superimposed each operating in respective, distinct frequency-bands. Frequency selective feedback compensation and adaptation beamforming may easily be realised, if at least beamforming in unit 12 as well as compensation in unit 18 are performed in frequency domain respectively in sub-bands. Beamforming is then realised at the frequencies f_{AFB} with minimum amplification at the specific angles α_{AFB}, whereas beamforming at other frequencies $f_{\overline{AFB}}$ is performed according to other needs. Consequently the adaptation rate of beamforming in unit 12 is only controlled by the gain of compensator unit 18 at the frequencies f_{AFB}.

[0130] Thus, even when beamforming $B_{\overline{AFB}}$ is switched off to minimum overall amplification, beamforming B_{AFB} may be maintained active to suppress feedback also in such "quiet" mode. Thereby, and with an eye on processing in frequency domain, in each sub-band, which is significant for AFB, the loop gain, as estimated in compensator unit 18, may be compared with a threshold value and adaptation rate control at C_{12R} is only established, if the instantaneous loop gain at least reaches such threshold. The control of the adaptation rate may then be lowered to practically zero, which means that beamforming is switched off for frequencies F_{AFB}. This establishes a hard on/off-switching of beamforming in the F_{AFB} frequency-range. In a further approach, such switching may be performed steadily which may be realised on the one hand by lowering the adaptation rate of B_{AFB} steadily and/or by reducing beamforming amplification of B_{AFB} steadily.

[0131] Due to the inventively improved suppression of acoustical feedback from the output of the output converter to the input of the input converter, there is reached additional stability of the device. The inter dependencies of vent tailoring at in-the-ear hearing devices and acoustical feedback problems is resolved to a significantly higher degree than was possible up to now when the device had the ability of adaptive beamforming.

1. A method for suppressing feedback between an acoustical output of an electrical/acoustical output converter arrangement and an acoustical input of an acoustical/electrical input converter arrangement of a hearing device, wherein

- acoustical signals impinging on the input converter arrangement are converted into a first electric signal by a controllably variable transfer characteristic which is dependent on the angle at which said at acoustical signals impinge on said input converter arrangement;

- said first electric signal is processed and a resulting signal is applied to the output converter arrangement;

- said feedback to be suppressed is compensated by a feedback compensating signal which is generated in dependency of the resulting signal and is fed back by a feedback signal path upstream said processing;

wherein further

- said electric feedback compensating signal is fed back to and superimposed upon the first electric signal and

- adaptation rate of said converting to variations of said transfer characteristic is controlled in dependency of the loop gain along said feedback signal path.

2. The method of claim 1, further comprising slowing down the adaptation rate of said converting with increasing loop gain along said feedback signal path.

3. The method of claims 1 or 2, further comprising minimising amplification of said transfer characteristic at one or more specific angles which accord to angles at which said feedback to be suppressed predominantly impinges on said input converter arrangement.

4. The method of one of claims 1 to 3, further comprising frequency selectively controlling said adaptation rate.

5. The method of one of claims 1 to 4, further comprising performing said converting in said first electric signal, and said processing along said feedback signal path in frequency domain and controlling said adaptation rate at selected frequencies in dependency of said loop gain at said selected frequencies.

6. The method of one of claims 1 to 5, further comprising minimizing amplification of said transfer characteristic at specific angles frequency selectively.

7. The method of one of claims 1 to 6, further comprising performing said converting into said first electric signal independently for frequencies present in said feedback to be suppressed and for frequencies substantially not present in said feedback to be suppressed.

8. The method of one of claims 1 to 7, further comprising performing said control of said adaptation rate selectively for frequencies present in said feedback to be suppressed, said control comprising switching said converting on and off for said frequencies present.

9. The method of claim 8, further comprising performing switching from on to off and/or vice versa steadily during a predetermined timespan.

10. The method of one of claims 1 to 9, said hearing device being a behind-the-ear or a in-the-ear hearing device.

11. The method of one of claims 1 to 10, said hearing device being a ear protection or a hearing improvement device.

12. A hearing device, comprising:

- an acoustical/electrical input converter arrangement and an adaptive beamformer unit, generating at an output an electric output signal dependent on acoustical signals impinging on said input converter arrangement and in dependency of angle at which said acoustical signals impinge, said beamformer unit having a first control input for varying beamforming characteristics
- a processing unit with an input operationally connected to the output of said beamformer unit and with an output operationally connected to an input of an electrical/acoustical output converter arrangement
- a feedback compensator unit, the input thereof being operationally connected to said input of said electrical/acoustical output converter arrangement, an output thereof being operationally connected to the input of said processing unit

and wherein further

- said beamformer unit has a second control input for adjusting adaptation rate,
- said output of said feedback compensator unit is operationally superimposed with the output of said beamformer unit,
- said feedback compensator unit has an output for a loop gain indicative signal, being operationally connected to said second control input of said beamformer unit.

13. The device of claim 12 being a behind-the-ear hearing device or an in-the-ear hearing device.

14. The device of one of claims 12 or 13, being a hearing protection device or a hearing improvement device.

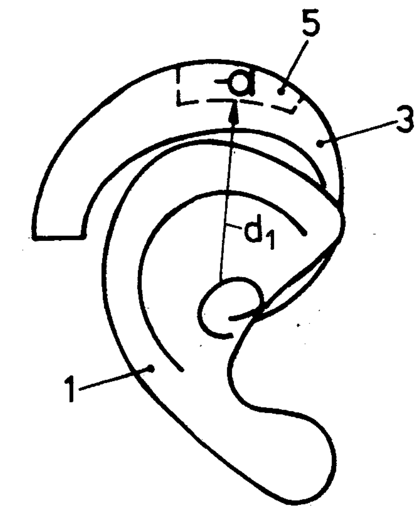


FIG.1

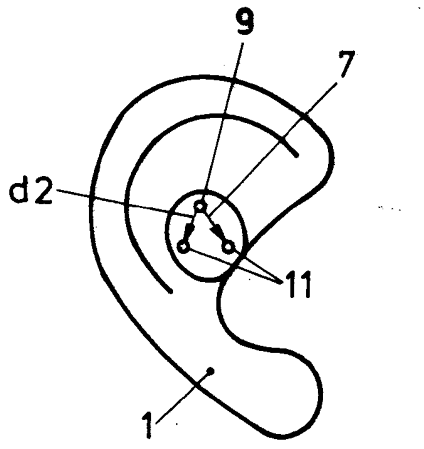


FIG.2

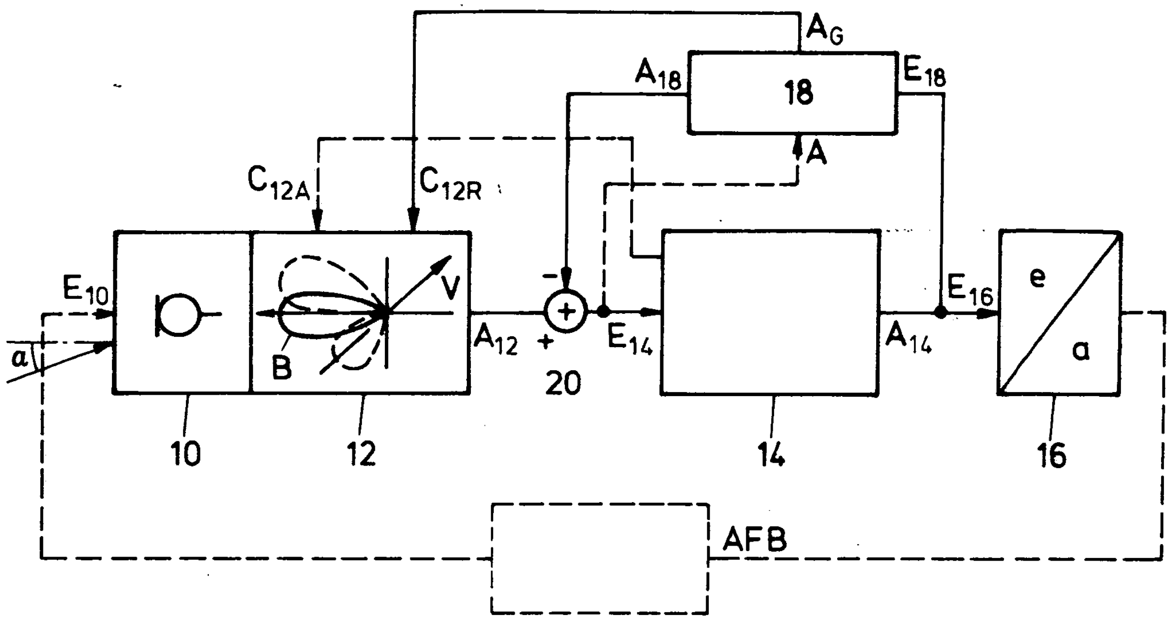


FIG. 3

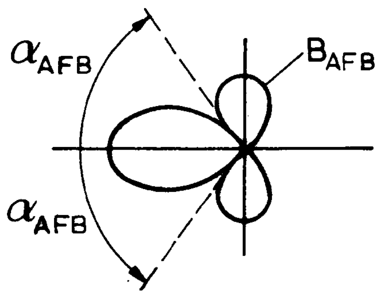


FIG. 4

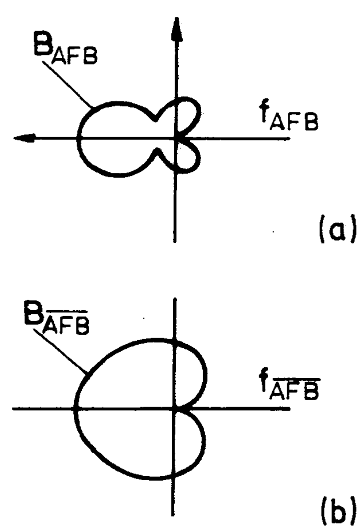


FIG. 5

Claims

1. A method for matching at least two acoustical to electrical converters, signals dependent on the electrical output signals of said converters being computed to result in a result signal and wherein the transfer characteristic between an acoustical signal impinging upon said at least two converters and said result signal is dependent on direction of arrival of said acoustical signal on said at least two converters, comprising the steps of matching said converters for acoustical signals in dependency of an impinging direction of arrival within a range determined before matching of direction of arrival upon said converters.

2. The method of claim 1, said transfer characteristic having a minimum for a value of direction of arrival and comprising the steps of matching said at least two converters for acoustical signals impinging within said range including said value.

3. The method of claim 1, wherein said dependent signals are first computed to result in a first of said result signals and are second computed to result in a second of said result signals and wherein a first transfer characteristic between an acoustical signal impinging upon said at least two converters and said first result signal is differently dependent on direction of arrival than a second transfer characteristic between said acoustical signal and said second result signal and wherein said matching is performed independently for said first and for said second computing.

4. The method of claim 1, further comprising performing said matching selectively in frequency bands determined before said matching.

5. The method of claim 1, further comprising performing an analog to digital and a time-domain to frequency-domain conversion downstream said converters.

6. The method of claim 1, wherein said result signal is operationally connected to an electric input of an electrical to acoustical converter comprising feeding back an electric feedback compensating signal dependent on an input signal to said electrical to acoustical converter and superimposing said fed back signal to said result signal, controlling adaptation rate of said matching in dependency of the loop gain along the feedback signal path of said electric feedback compensating signal.

7. The method of claim 1, further comprising performing said matching with a matching time constant τ for which there is valid:

$$0 < \tau \leq 5 \text{ sec.}$$

8. The method of claim 1, wherein

$$0 < \tau \leq 1 \text{ sec.}$$

is valid.

9. The method of claim 8, wherein

$$0 < \tau \leq 100 \text{ msec.}$$

is valid.

10. A method for matching at least two acoustical to electrical converters, signals dependent on the electrical output signals of said converters being computed to result in a result signal and wherein the transfer characteristic between an acoustical signal impinging upon said at least two converters and said result signal is dependent on direction of arrival of said acoustical signal on said at least two converters comprising the step of matching said converters with a matching time constant τ:

$$0 < \tau \leq 5 \text{ sec.}$$

11. The method of claim 10, wherein there is valid:

$$0 < \tau \leq 1 \text{ sec.}$$

12. The method of claim 11, wherein there is valid:

$$0 < \tau \leq 100 \text{ msec.}$$

13. A beamforming device comprising at least two acoustical to electrical converters and at least one computing unit, the electrical outputs of said converters being operationally connected via a matching unit to inputs of said at least one computing unit, the output of said beamforming device being operationally connected to the output of said at least one computing unit, said computing unit generating a signal indicative of direction of arrival of an acoustical signal impinging on said at least two converters, a matching control unit generating a matching control signal operationally connected to a control input of said matching unit, said signal indicative of direction being operationally connected to a control input of said matching control unit, said matching unit comprising at least two inputs operationally connected to the outputs of said at least two converters upstream or downstream said matching unit.

14. The beamforming device of claim 13, further comprising an electrical to acoustical converter, the output of said computing unit being operationally connected to an input of said electrical to acoustical converter, further comprising a feedback compensator unit, the input thereof being operationally connected to said input of said electrical to acoustical converter, an output thereof being operationally connected and superimposed to the output of said computing unit, said feedback compensator unit having an output for a loop gain indicative signal being operationally connected to a control input of said matching control unit.

15. The device of claim 13 being part of a hearing device.

16. The device of claim 15, wherein said hearing device is an outside-the-ear hearing device or an in-the-ear hearing device.

17. The device of claim 15, wherein said hearing device is a hearing improvement device, a hearing aid device or a hearing protection device.

18. The beamforming device according to claim 13, said matching control unit generating said matching control signal to control said matching with a matching time constant τ for which there is valid:

$$0 < \tau \leq 5 \text{ sec.}$$

19. The device of claim 18, wherein there is valid:

$$0 < \tau \leq 1 \text{ sec.}$$

20. The device of claim 19, wherein there is valid:

$$0 < \tau \leq 100 \text{ msec.}$$

21. A beamforming device comprising at least two acoustical to electrical converters and at least one computing unit, the electrical outputs of said converters being operationally connected via a matching unit to inputs of said at least one computing unit, the output of said beamforming device being operationally connected to the output of said at least one computing unit, a matching control unit generating a matching control signal operationally connected to a control input of said matching unit, said matching unit comprising at least two inputs operationally connected to the outputs of said at least two converters upstream or downstream said matching unit, said matching control unit generating said matching control signal to control said matching with matching time constant τ for which there is valid:

$$0 < \tau \leq 5 \text{ sec.}$$

22. The device of claim 21, wherein there is valid:

$$0 < \tau \leq 1 \text{ sec.}$$

23. The device of claim 22, wherein there is valid:

$$0 < \tau \leq 100 \text{ msec.}$$

24. A beamforming device comprising at least two acoustical to electrical converters and at least one computing unit, the electrical outputs of said converters being operationally connected to inputs of said at least one computing unit, the output of said beamforming device being operationally connected to the output of said at least one computing unit, further comprising means for generating a signal indicative of direction of arrival of an acoustical signal impinging on said converters, further comprising means for performing matching of said at least two acoustical to electrical converters, said means for matching being controlled by said signal indicative of said direction of arrival.

25. A beamforming device comprising at least two acoustical to electrical converters and at least one computing unit, further comprising means for matching said at least two acoustical to electrical converters, said means for matching operating with a matching time constant τ for which there is valid:

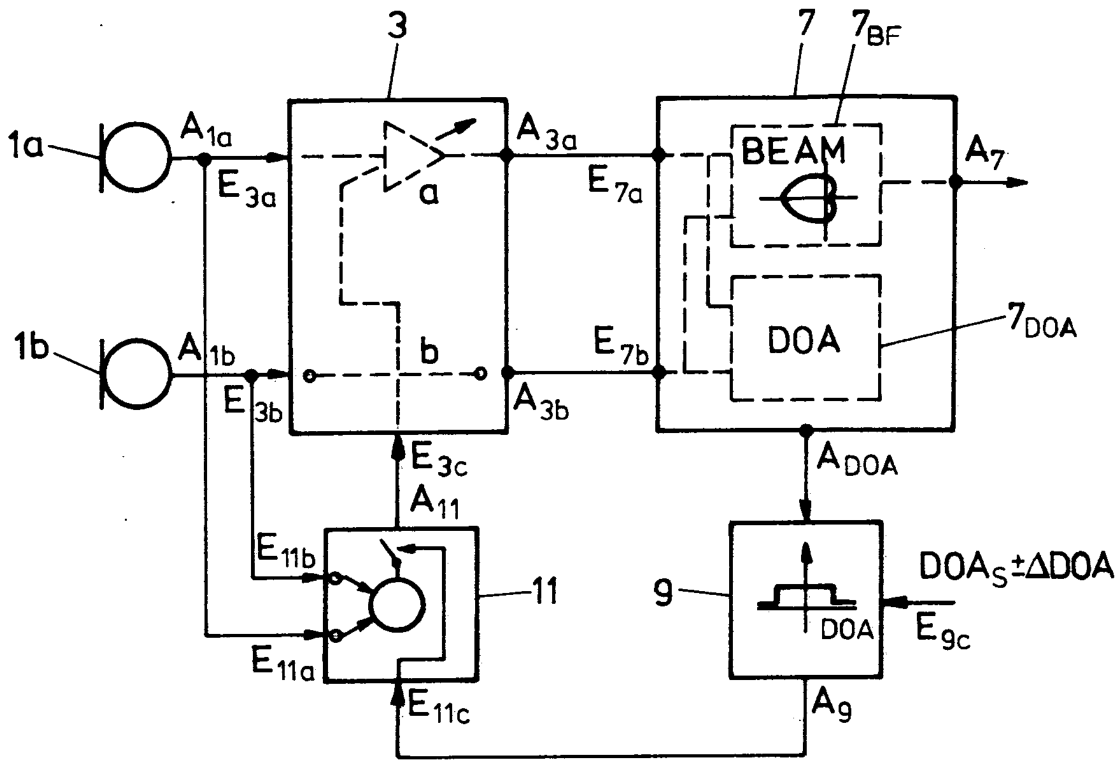
$$0 < \tau \leq 5 \text{ sec.}$$

26. The device of claim 25, wherein there is valid:

$$0 < \tau \leq 1 \text{ sec.}$$

27. The device of claim 26, wherein there is valid:

$$0 < \tau \leq 100 \text{ msec.}$$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BEAM
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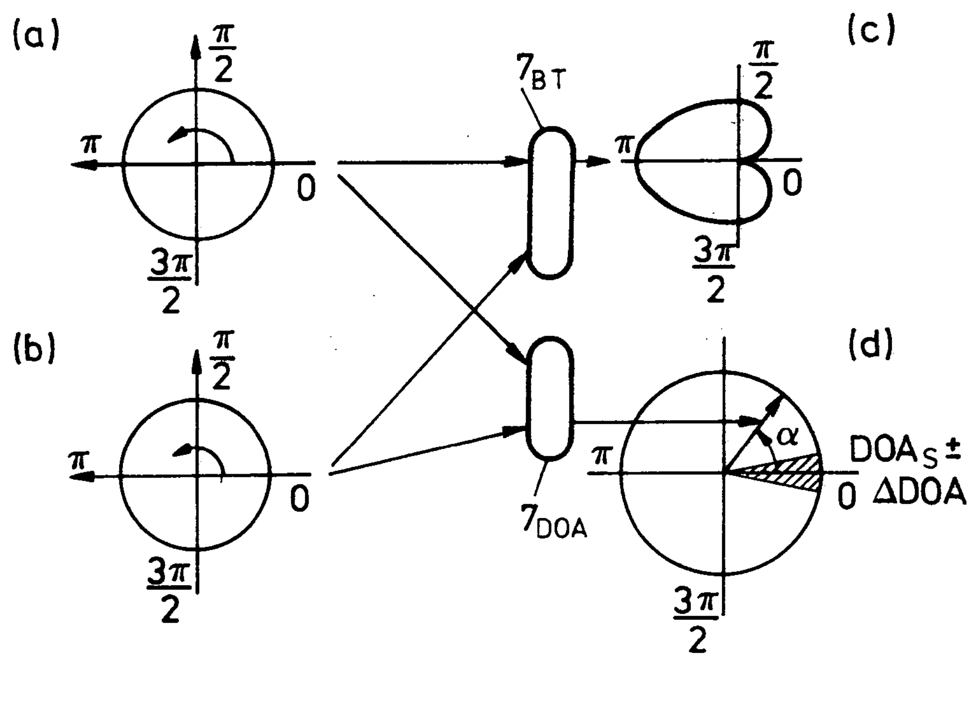
A_{1b}
E_{3b}
b
E_{7b}
A_{3b}
DOA
7_{DOA}
E_{3c}
A_{11}
A_{DOA}
E_{11b}
11
9
DOA
$DOA_S \pm \Delta DOA$
E_{9c}
E_{11a}
E_{11c}
A_9

FIG.1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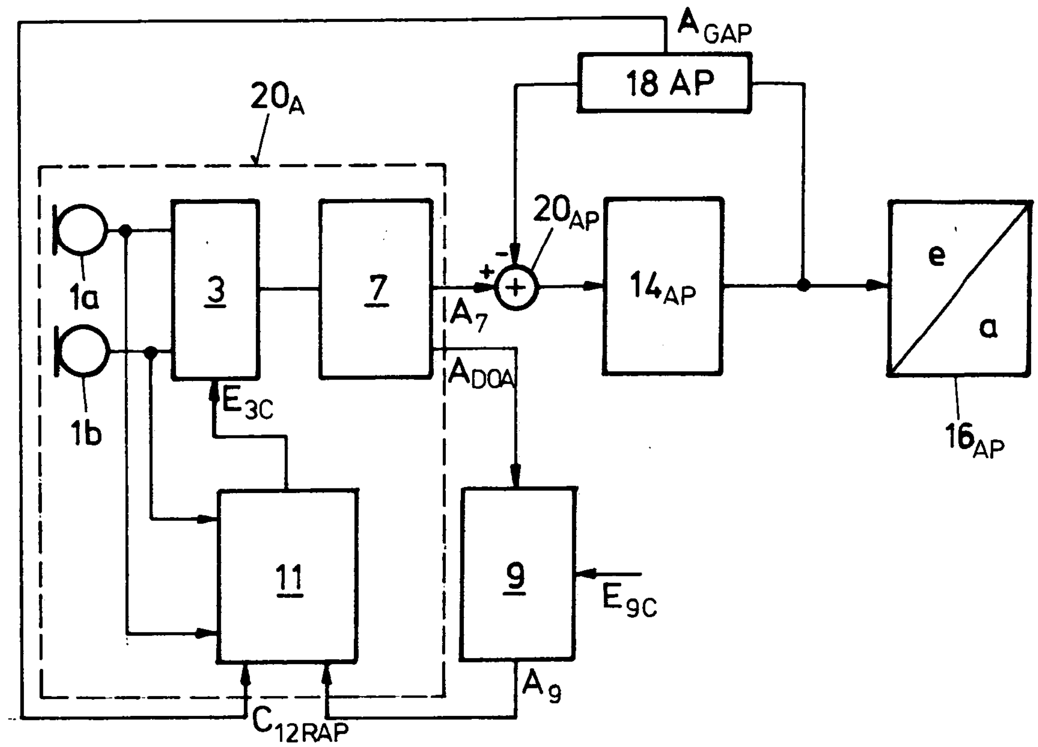
(c)
$\frac{\pi}{2}$
7_{BT}
π
0
$\frac{3\pi}{2}$
(b)
(d)
α
$DOA_S \pm$
ΔDOA
7_{DOA}

FIG. 2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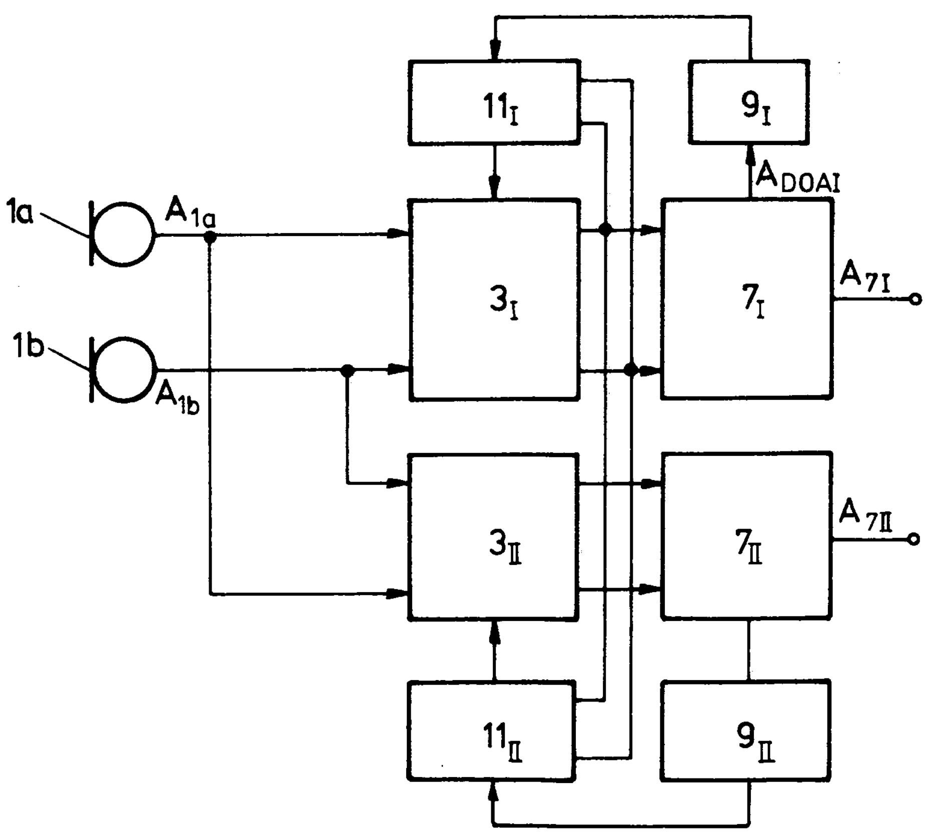
20A
20AP
-
+
3
7
14AP
e
a
1a
1b
A7
ADOA
E3C
11
9
E9C
A9
C12RAP
16AP

FIG.3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ADOAI
1a
A1a
3I
7I
A7I
1b
A1b
3II
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A7II
11II
9II