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(54) **Model selection of acoustic conditions for active noise control**

(57) It is proposed active noise control method for reducing the amount of noise in a local zone (CZ) comprising capturing at least one audio signal (e, x) inside an area (A) including at least the local zone and generating an anti-noise signal (y) which is function of this at least one audio signal and from a model of the acoustic characteristics of at least a part of the area, wherein this model is selected among a set (B) of predetermined models in accordance with at least one physical measurement (s(n)) representative of these acoustic characteristics.

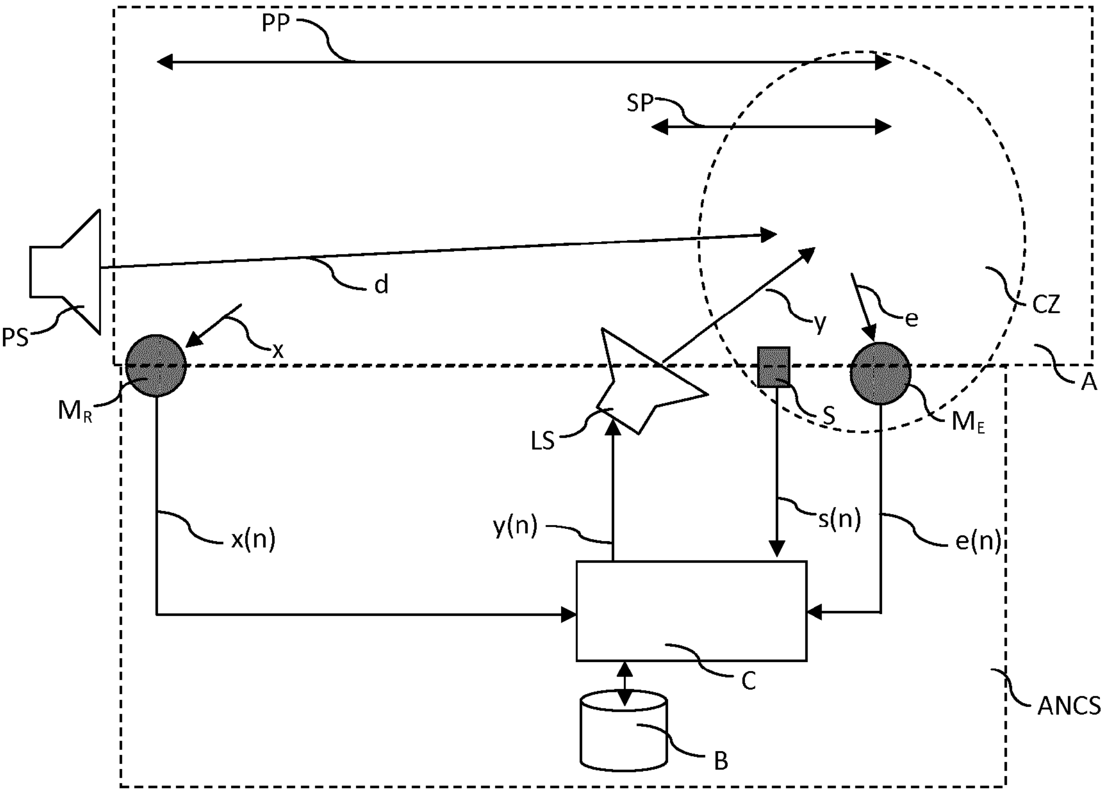

Fig. 2

EP 2 701 143 A1

Description

FIELD OF THE INVENTION

[0001] The invention relates to the field of active noise control (ANC), and more especially to the application of ANC to mobile communication terminals.

BACKGROUND OF THE INVENTION

[0002] Active Noise Control, also known as Active Noise Cancellation or Active Noise Reduction (ANR), is a set of techniques based on the principle of destructive interferences between sound fields, in order to reduce or cancel unwanted sound. The unwanted sound components (generally, noise) are locally cancelled or attenuated by superposing a sound wave with the same amplitude but with an inverted phase. This sound wave is often referred to as "anti-noise".

[0003] They are two basic approaches to the active noise control: the feed-forward control and the feedback control. Approaches mixing feed-forward and feedback control are also possible. Active control can also be adaptive or static.

[0004] The figure 1 illustrates these two approaches.

[0005] A primary source PS generates an unwanted sound. This sound can be noise, or any other signal. This source has been represented as a sole loudspeaker only for the clarity of the figure. In real situations, the unwanted sound can be of various natures (motors, street rumors, etc.) and comes from multiple sources.

[0006] At a distant of this primary source, a loudspeaker LS has been introduced to generate the anti-noise signal. Its aim is to cancel or at least dramatically reduce the unwanted signal in a cancellation zone CZ.

[0007] An error microphone M_E is placed within the cancellation zone CZ, in the vicinity of the loudspeaker LS in order to control the signal resulting of the emission of the anti-noise signal and the unwanted signal. It aims in capturing what is really heard by the human ear within the cancellation zone. In the ideal case, the error microphone M_E measures a null signal.

[0008] The path between the loudspeaker LS and the error microphone M_E is called the secondary path SP.

[0009] The simplest implementation of the feedback approach consists for the Active Noise (AN) Controller ANCS in capturing the measured signal e(n) and in re-injecting in the loudspeaker LS a signal y(n) which is a copy of e(n) with an inversion of the phase and a adaption of the amplitude.

[0010] The feed-forward approach is based on a reference microphone M_R which capture a signal far enough from the loudspeaker LS so as the captured signal is not perturbed by the introduced signal. In other words, the signal x(n) captured by the reference microphone M_R is representative of the unwanted signal.

[0011] The feed-forward approach is further based on an a-priori knowledge of the transfer function of the channel linking the primary source and the cancellation zone CZ. This path is equivalent to the path between the reference microphone M_R and the error microphone M_E, which is usually referred to as "primary path" PP.

[0012] Knowing the transfer function of the primary path PP, the AN controller can use a filter modeling this transfer function so as to determine the re-injected signal y(n) as a function of the signal x(n) measured by the reference microphone M_R.

[0013] In general, however, the active noise controllers make use of both approach and output a re-injected signal y(n) which is function of both the signal x(n) measured by the reference microphone M_R and the signal e(n) measured by the error microphone M_E. The AN Controller ANCS can implement various algorithm to take into account both sources of information for generating the most optimized corrective signal y(n).

[0014] For generating an efficient cancellation signal y(n), the AN Controller ANCS should also take into account the transfer function of the secondary path SP. Whatever the chosen approach (feed-forward, feedback, or mixed), the secondary path SP should be modelized and the re-injected signal y(n) should depend on this model.

[0015] This secondary path SP includes the acoustic area between the loudspeaker LS and the error microphone M_E which should be as close as possible of the user's ear. It also includes the transfer functions from the transducers and the audio convertors. Therefore, its model depends on the physical arrangement in which the Active Noise Control is used.

[0016] In some cases, like for instance active headphones, the physical arrangement is predetermined and fixed. The model can be preset and the above-described techniques works finely. It will be the same in other situations where the secondary path linking the loudspeaker and the error microphone does not change substantially over time.

[0017] However, there is now a demand for applying active noise control in conditions where this secondary path can change substantially, in terms of amplitude, phase and rapidity.

[0018] This is for instance the case for mobile communication terminals (or mobile phones).

[0019] There is a need to take benefit of the ANC advantages to reduce or even cancel the noise around a user of a mobile phone, in order to improve the quality of the sound perceived by the user and emitted by the speech loudspeaker of the mobile phone. This feature improves the intelligibility of the speech during a phone call in a noisy environment, e.g. without increasing the volume of the speech loudspeaker.

[0020] However, the user can hold the mobile terminal in various positions and press it more or less tightly on his/her ear. Also, he or she can also put it on a table, or move it from one ear to the other, etc. These variations in the acoustic characteristics of the communication channel between the loudspeaker and user's ear makes it impossible to rely on a fixed model of the secondary path SP.

[0021] A solution to this issue consists in estimating the model of secondary path SP in real-time, so that the AN Controller generates the corrective signal y(n) according to parameters adapted to current acoustic conditions of the secondary path.

[0022] On-line secondary path techniques exist but are difficult to implement in the context of a mobile communication terminal.

[0023] They require continuous monitoring of the secondary path SP to generate in real time parameters needed for the AN Controller ANCS to generate the appropriate corrective signal y(n).

[0024] Other techniques are based on the introduction of an additive signal (noise or sweep) as an excitation signal for identifying the secondary path. But such techniques are not transparent for the user, since this added signal can be heard, and should there be also avoided.

[0025] There is a need for a solution permitting to efficiently apply active noise control in the context of mobile communication terminals or, of any device in which the secondary path may evolve over time.

SUMMARY OF THE INVENTION

[0026] An object of embodiments of the present invention is to alleviate at least partly the above mentioned drawbacks

[0027] This is achieved with an active noise control method for reducing the amount of noise in a local zone comprising

- capturing at least one audio signal inside an area including at least said local zone and
- generating an anti-noise signal which is function of said at least one audio signal and from a model of the acoustic characteristics of at least a part of this area,
- wherein this model is selected among a set of predetermined models, in accordance with at least one physical measurement representative of the acoustic characteristics.

[0028] According to embodiments of the invention, the physical measurement is distinct of the at least one audio signal.

[0029] The at least a part of said area can be a secondary path, defined as the space between the loudspeaker and an error microphone situated inside the local zone.

[0030] The at least one physical measurement can comprise an electrical impedance measured on this error microphone.

[0031] The anti-noise signal can be generated by a loudspeaker.

[0032] The at least one physical measurement can comprise an indication of proximity of an object of the loudspeaker.

[0033] The at least a part of said area can be a primary path, defined as the space between an error microphone (M_E) situated inside said local zone (CZ), and a reference microphone situated outside the local zone, e.g. far outside the local zone.

[0034] The at least one physical measurement can comprise an estimation of the primary path.

[0035] Another object of the invention is an active noise control system for reducing the amount of noise in a local zone, comprising:

- A microphone for capturing at least one audio signal inside an area including at least this local zone,
- A loudspeaker for generating an anti-noise signal which is function of the at least one audio signal and from a model of the acoustic characteristics of at least a part of this area,
- A sensor for capturing at least one physical measurement representative of the acoustic characteristics;
- a memory for storing a set of predetermined models,
- a classifier for selecting the model among this set of predetermined models, in accordance with this at least one physical measurement.

[0036] According to embodiments of the invention, the physical measurement can be distinct of said at least one audio signal.

[0037] The at least a part of said area can be a secondary path, defined as the space between said loudspeaker and an error microphone situated inside said local zone.

[0038] The at least one physical measurement can comprise an electrical impedance measured on the error microphone.

[0039] The anti-noise signal can be generated by a loudspeaker.

[0040] The at least one physical measurement can comprise an indication of proximity of an object of the loudspeaker.

[0041] Another object of the invention is a computer program product comprising a computer readable medium, having thereon a computer program comprising program instructions, the computer program being loadable into a data-processing unit and adapted to cause execution of the method previously defined when the computer program is run by the data-processing unit.

BRIEF DESCRIPTION OF THE DRAWINGS

[0042]

Fig. 1, already described, illustrates a functional high-level architecture of the active noise control of the state-of-the-art.

Fig. 2 shows a functional high-level architecture of an active noise control system according to an embodiment of the invention.

Fig. 3, 4 and 5 show three block diagrams of three example embodiments of the invention.

Figure 6 illustrates an application of the invention to a mobile phone.

DETAILED DESCRIPTION OF EMBODIMENTS OF THE INVENTION

[0043] An active noise control system ANCS is depicted on the figure 2. It is located in front of an area A which comprises a cancellation zone CZ.

[0044] The cancellation zone CZ (or local zone) is the portion of space in which the active noise control is wished to produce effects. As it is a continuous effect, the border of the cancellation zone CZ can be defined by a threshold: the effect of the cancellation/reduction of the zone inside the cancellation zone CZ is assumed to be above the threshold.

[0045] The cancellation zone CZ corresponds to the usage of the Active Noise Control. For instance, if the device in which the ANC is embedded is a pair of headphone, then the cancellation zone can correspond to the closed space between the ear and the headphone. If the device is a mobile communication terminal (mobile phone), then the cancellation zone is a more fuzzy space in which the user is expected to put his/her ear and the device itself.

[0046] In addition to the active noise control system ANCS, the device usually comprises several other elements, e.g. a loudspeaker SLS adapted to emit sounds related to a communication channel. In the case of a mobile phone, this loudspeaker emits the voice of the other party(ies) of the communication session. It can also emit music, vocal messages or any other audio media.

[0047] In the figure 6, this loudspeaker SLS is depicted as distinct of the loudspeaker LS comprised in the active noise control system and devoted to the generation of anti-noise signal. This arrangement is mainly done for the purpose of the explanations by clearly distinguishing the functions of each element. In real-life implementations, both loudspeakers are implemented as a single one; this loudspeaker will emit a signal mixing the audio media and the anti-noise signal. The mixing can be done in the digital or analog electric worlds.

[0048] Still in the example of figure 6, it also comprises a signal microphone Ms adapted to capture the voice of the user (or any other audio signal), especially for transmission over a communication network toward to other party(ies) of a communication session.

[0049] The mobile communication terminal T further comprises a screen D. According to the state of the art, this screen enables the user to see information provided by the terminal, but also inputs information and commands, by tactile features.

[0050] It also comprises an error microphone M_E, a reference microphone M_R and a sensor S. As it will be explained with more details later, the reference microphone M_R aims at capturing the ambient noise. Therefore, it should be placed far enough of the loudspeakers SLS and LS. It may be chosen to put it also at distance of the user's mouth.

[0051] In the figure, it is depicted on the side of the mobile phone T, but other locations are possible like, for instance, its rear side.

[0052] The figure 6 should be considered as a particular embodiment of the invention. The elements can be arranged in many different ways and some may be even omitted without departing from the scope of the invention.

[0053] This is notably the case of the reference microphone M_R and/or the error microphone M_E which are useful only according to certain embodiments of the active noise control system. In general, one among both of them is however required.

[0054] Turning back to the figure 2, elements of the device which are not comprised in the active noise control system ANCS are not depicted. By contrast, the reference microphone M_R, the error microphone M_E, the sensor S and the loudspeaker LS for generating an anti-noise signal y are depicted in this figure.

[0055] The noise d which amount is to be reduced is shown as emitted by a source PS. However, the source of this noise may not be unique. It can even be uncountable, as in the case of a rumors made of hundreds of noises which may even be of different natures. For example, in the streets, a mobile phone can be surrounded by noises coming from traffic, people's voices, music coming out from shops, etc.

[0056] For the purpose of clarity, all these sources will be further referred to as a single "virtual source" PS, generating a noise d, as the invention is independent of the number, location and nature of the noise sources.

[0057] The reference microphone M_R is located close to the virtual source PS. Preferably, it should be far enough of the loudspeaker LS to capture an audio signal x that is assumed to be a good estimate for the noise d (i.e. which is not significantly perturbed by the anti-noise signal y or any other signal emitted by a signal loudspeaker SLS, not represented).

[0058] The error microphone M_E is located close to the loudspeaker LS. It aims in capturing an error signal e, inside the cancellation area CZ, which results from the superposition of the noise signal d and the anti-noise signal generated by the loudspeaker LS. It thus provides an objective measurement of the performances of the active noise control system ANCS (i.e. how well the anti-noise signal y fully compensates for the noise signal d).

[0059] The error microphone M_E and the reference microphone M_R both capture an audio signal, respectively e, x, inside the area A. According to the invention, only one of them can be present, so as to capture only one for the audio signals e, x. According to variants of the invention, both microphones can be present, so as to capture and take benefit of these two audio signals e, x.

[0060] The captured audio signals e, x are transformed by the microphones in electric digital signals, respectively e(n), x(n).

[0061] The anti-noise signal y generated by the loudspeaker LS is function of these one or two (or, potentially more) audio signals e, y, and also of a model of the acoustic characteristics of a part of this area A. The part can be the totality of this area A.

[0062] In embodiments of the invention, this part of the area A is a secondary path SP, which includes the acoustic path between the loudspeaker LS and the error microphone M_E situated inside the cancellation zone CZ.

[0063] In other embodiments of the invention, this part is a primary path PP defined as the space between the error microphone M_E and the reference microphone M_R.

[0064] Furthermore, the active noise control system ACNS comprises a sensor S for capturing at least one physical measurement s(n) representative of the acoustic characteristics of this part of the area A.

[0065] This physical measurement s(n), as well as the electric signals e(n), x(n) corresponding to the error and reference audio signals e, x, are transmitted to a classifier C which configures the active noise control system ANCS. The anti-noise electric signal y(n) is transmitted to the loudspeaker LS which reacts by generating the corresponding audio signal y.

[0066] In order to output this anti-noise signal y(n), the classifier selects a model of the acoustic condition of the considered part(s) of the area A, among a set B of predetermined models. This selection is done according to at least the physical measurement s(n).

[0067] The predetermined models can be designed offline. Therefore, even a precise tuning of them does not harm the responsiveness of the classifier C. According to the invention, thus, the anti-noise signal y(n) can dynamically follow any variations of the input signals e(n), x(n), s(n). The anti-noise signal y can adapt in real-time to the changes in the acoustic conditions of the considered part(s) of the area A, in a transparent way for the user: no audio artifacts are heard even in cases of fast and important changes like when the user moves the mobile phone from the ear to put it on a table, etc.

[0068] The classifier C can be implemented in various ways without departing from the scope of the invention.

[0069] One possible embodiment consists in conforming to front-end/back-end split architecture.

[0070] The front end operates on the physical measurement s(n), extracts features it and group the features into a vector sent to the back end.

[0071] Different types of classifiers are possible.

[0072] The usage of a threshold is the simplest solution to discriminate between two classes of acoustic conditions. For instance, it has been observed, on a phone mockup and an artificial ear, that a threshold criterion applied at 1800 Hz on an estimated primary path transfer function discriminates two classes of acoustic conditions:

- One with the ear close to the phone mockup;
- One with a leak between the ear and the phone mockup.

[0073] According to this threshold-based discrimination, the classifier C can output two different signals y(n) to command the loudspeaker LS. This way, the presence or absence of the ear in the cancellation zone CZ is detected in real-time and can dynamically command different anti-noise signal y to adapt the situation.

[0074] The threshold depends on the physical parameters of the microphone, the shape of the phone etc. and should be setup by a calibration of the Active Noise Control System, or by statistical analysis of a database of measurements, for examples.

[0075] Other embodiments may make use of techniques based on Support Vector Machine (SVM). SVM is a well-known classification family of algorithms. The original algorithm was described in the article "Support Vector Networks" of Corinna Cortès and Vladimir Vapnik, in Machine Learning 20, 1995, Springer.

[0076] A supervised learning procedure can associate extracted features with acoustic conditions and estimated secondary path. Learning can be performed offline. At runtime, the SVM classifier is fast and efficient: its decisions rely mainly on the sign of different scalar products in the feature space.

[0077] The output of the classifier is not restricted to a discrete enumeration of recognized acoustic conditions: weighting or confidence estimates obtained from the recognition engine can be used as well in order to enable interpolation between acoustic conditions.

[0078] The physical measurements s(n) can be of various types also.

[0079] According to a preferred embodiment of the invention, the physical measurement is distinct of the audio signals.

[0080] This enables to get rid of the noisy nature of the audio signals and to base the classification on "clean" signal. The classification can then been done with more accuracy and more reliability. The process can then be more stable and adapt to a wider range of situation.

[0081] The physical measurement can be a proximity measurement provided by a proximity sensor S. As proximity sensors are commonly implemented in recent mobile phones, such a solution will not add any manufacturing costs.

[0082] The physical measurement can be a pressure measurement provided by a mechanical pressure sensor. This sensor can give an indication of the presence of the ear behind the loudspeaker LS and also discriminates between classes where the mobile phone is hold on the ear and where it is hold far from the ear.

[0083] The physical measurement can be a current intensity signals coming from loudspeaker LS. Indeed, it has been observed that the current intensity flowing through a loudspeaker (and its impedance) depends on the acoustic load installed in front of it. Laboratory measurements' demonstrate a strong relationship between the impedance (or current intensity), the type of obstacle (an artificial ear, in the laboratory experiments) and its relative location in front of the loudspeaker.

[0084] In particular, the location and the shape of the resonances are correlated with the acoustic conditions. There is no doubt that this correlation can be used in order to recognize different acoustic conditions and secondary path transfer functions.

[0085] Other physical measures are possible. Also, a combination of physical measurements can be used as inputs of the classifiers C, so as to provide more robustness to the system.

[0086] The figures 3, 4 and 5 illustrate some particular embodiments of the invention. Some aspects of the invention that have been briefly evocated here above will be described more clearly in view of these particular embodiments.

[0087] The figure 3 illustrates an embodiment based on a feedback ANC and current intensity measurements.

[0088] The error microphone 301 captures an audio signal in the acoustic space and transform it into an electric signal through a transfer function M(s). This signal can modulate an electric carrier by a modulator 302 and is digitalized by an ADC (Analog-to-Digital Convertor) 303.

[0089] This signal goes then through a filter 304 associated with a filter function w(z), provided by a filter selection function 311. This filter minimizes the sensibility of the closed system (i.e. the rejection gain of the external noise) given the model of the acoustic conditions in front of the error microphone 301. This microphone being close to the loudspeaker, these acoustic conditions are identical or substantially similar to the ones in front of the error microphone.

[0090] The filtered signal is then transmitted to a DAC 305 (Digital-to-Analog Convertor).

[0091] It goes through a current sensing function 306 which extracts from the signal the current intensity, without modifying substantially the signal. Such an extraction can be done by a resistance. The signal sent to the loudspeaker 307 which transform the electric signal to an acoustic signal by a transfer function L(s).

[0092] The acoustic signal then flows through the secondary path 308, to which a transfer function S(s) can be associated. This transfer function S(s) represents attenuations, echoes and other acoustic phenomena associated to geometrical features of the secondary path, obstacles, etc.

[0093] The resulting audio signal is superimposed with other sounds (e.g. noise) in a virtual function 309.

[0094] This open loop function is unknown. The filter w(z) still minimizes the sensibility function of the closed loop system given by the estimated model of the open loop function. The filter w(z) is selected by taking as inputs the estimated impedance, which has been estimated given the signal digitalized by the ADC.

[0095] As earlier explained, this current intensity is correlated by the acoustic conditions in the vicinity of the loudspeaker 307.

[0096] According to the value of this intensity, an appropriate filter can be selected among a database of available filters 312. Classification techniques mentioned earlier can be used for determining the most appropriate filter to the measured acoustic conditions.

[0097] The figure 4 illustrates an embodiment based on a feedback ANC and a filter selection based on a primary path estimation. The block diagram of the figure 4 is similar to the block diagram of the figure 3, with the differences of the blocks related to the capture of information used for selecting the appropriate filter.

[0098] The error microphone 401 captures an audio signal in the acoustic space and transform it into an electric signal through a transfer function M(s). This signal can modulate an electric carrier by a modulator 402 and is digitalized by an ADC (Analog-to-Digital Convertor) 403.

[0099] This signal goes then through a filter 404 associated with a filter function w(z), provided by a filter selection function 411. This filter minimizes the sensibility of the closed system (i.e. the rejection gain of the external noise) given the model of the acoustic conditions in front of the loudspeaker 407.

[0100] The filtered signal is then transmitted to a DAC 405 (Digital-to-Analog Convertor), and then to the loudspeaker 307 which transform the electric signal to an acoustic signal by a transfer function L(s).

[0101] The acoustic signal then flows through the secondary path 408, to which a transfer function S(s) can be associated. This transfer function S(s) represents attenuations, echoes and other acoustic phenomena associated to geometrical features of the secondary path, obstacles, etc.

[0102] The resulting audio signal is superimposed with other sounds (e.g. noise) in a virtual function 409.

[0103] This open loop function is unknown. The filter w(z) still minimizes the sensibility function of the closed loop system given by the estimated model of the primary path (obtained from the adaptive feed-forward version of the ANC algorithm) or the estimation of the path between the output node of the ANC 410 and the input node of the DAC 405. The reference microphone 413 has a transfer function M(s). This transfer function may be similar or different of the one of the error microphone 401.

[0104] The estimation of the primary path, which is representative of the acoustic conditions of the primary path is then used as a criterion to select the most appropriate filter among a database of available filters 412. Classification techniques mentioned earlier can be used for determining the most appropriate filter to the measured acoustic conditions.

[0105] The features than can be extracted from the estimated acoustic conditions of the primary path (i.e. the space between the error microphone and the reference microphone). These acoustic conditions are directly related to the acoustic conditions of the pinna and to the actual secondary path transfer function; Therefore, the transfer function w(z) can be selected on this basis.

[0106] The justification for this correlation is intuitive: when the phone device seals the ear pinna without acoustic leak, the acoustic path between the error microphone and the reference microphone is cut and the primary path is attenuated when compared with a configuration without such obstacle or with acoustic leaks.

[0107] It has been earlier mentioned that on a phone mockup with an artificial ear, the maximal attenuation was observed at 1800 Hz. At this frequency, the power estimation of the primary path can be used as a relevant feature.

[0108] The figure 5 illustrates an embodiment based on an adaptive feed-forward model and filter selection based on an estimation of the current intensity. Several models for adaptive feed-forward approach exist, and the FxLMS feed-forward implementation of the invention should not be understood as limited to this particular functional architecture.

[0109] More information about this particular architecture can be found, for instance, in the diploma thesis "Active noise Control" of Aleksandar Milosevic and Urs Schaufelberger, University of Rapperwil, December 14, 2005.

[0110] The error microphone 501 captures an audio signal in the acoustic space and transform it into an electric signal through a transfer function M(s). This signal can modulate an electric carrier by a modulator 502 and is digitalized by an ADC (Analog-to-Digital Convertor) 503.

[0111] This digital signal is sent transmitted to a LMS block 504 (for Least Mean Square). The output of the LMS block corresponds to the coefficients of the FIR filter 505 associated with a time-varying transfer function $w_n(z)$, which is fed by a digital signal coming from the reference microphone 512.

[0112] The adaptive active noise control system ANCS is designed so as to converge as fast as possible, thanks to the LMS block 504 set before it. According to the literature, the filter $w_n(z)$ converges even in case of large variations, provided that the phase error on the secondary path is less than 90°.

[0113] This signal goes then from this filter 505 to a DAC 506and then to the loudspeaker 508 which transform the electric signal to an acoustic signal by a transfer function L(s). In-between, an current sensing block 507 enables to extract a measurement of the current intensity without modifying substantially the signal. The impedance which has been estimated given the signal digitalized by the ADC 515 and the input of the DAC 506 is used as inputs for a filter selection block 516.

[0114] As explained earlier in connection with figure 3, the most appropriate filter S' (z) is selected according to this measurement among a set of available filters 517. This filter S'(z) aims in estimating the transfer function S(s) associated with the secondary path. The effect of the secondary path can then be compensated.

[0115] This filter S'(z) has as input the digitalized measurement of a signal coming from the reference microphone 512. As earlier explained in connection with the figure 4, the audio signal captured by the microphone is transformed into an electric signal by a transfer function $M_R(s)$. This function can be the same of the transfer function $M_E(s)$ of the error microphone or different. It is then digitalized by the ADC 513 and then filtered by the filter 514. The filtered signal is then used as input of the LMS 504, described before.

[0116] In the acoustic domain, the audio signal produced by the loudspeaker 508 is sur-imposed (509) with the noise flowing through the primary path 511. This primary path is modelized by a transfer function P(s).

[0117] The invention has been described with reference to preferred embodiments. However, many variations are possible within the scope of the invention.

Claims

1. An active noise control method for reducing the amount of noise in a local zone (CZ) comprising capturing at least one audio signal (e, x) inside an area (A) including at least said local zone and generating an anti-noise signal (y) which is function of said at least one audio signal and from a model of the acoustic characteristics of at least a part of said area, wherein said model is selected among a set (B) of predetermined models, in accordance with at least one physical measurement (s(n)) representative of said acoustic characteristics.

2. The method of claim 1, wherein said physical measurement is distinct of said at least one audio signal.

3. The method of any of claim 1 or 2, wherein said at least a part of said area is a secondary path, defined as the space between said loudspeaker and an error microphone (M_E) situated inside said local zone (CZ).

4. The method of claim 3, wherein said at least one physical measurement comprises an electrical impedance measured on said error microphone (M_E).

5. The method of any of claims 1 to 4, wherein said anti-noise signal is generated by a loudspeaker (LS).

6. The method of claim 5, wherein said at least one physical measurement comprises an indication of proximity of an object of said loudspeaker.

7. The method according to any of previous claims, wherein said at least a part of said area is a primary path, defined as the space between an error microphone (M_E) situated inside said local zone (CZ), and a reference microphone situated outside said local zone.

8. The method according to the previous claim, wherein said at least one physical measurement comprises

an estimation of the primary path.

9. An active noise control system (ANCS) for reducing the amount of noise in a local zone (CZ), comprising:

- A microphone (M_E, M_R) for capturing at least one audio signal (e, x) inside an area (A) including at least said local zone,
- A loudspeaker (LS) for generating an anti-noise signal (y) which is function of said at least one audio signal and from a model of the acoustic characteristics of at least a part of said area,
- A sensor (S) for capturing at least one physical measurement (s(n)) representative of said acoustic characteristics;
- a memory for storing a set (B) of predetermined models,
- a classifier (C) for selecting said model among said set (B) of predetermined models, in accordance with said at least one physical measurement.

10. The active noise control system of claim 9, wherein said physical measurement is distinct of said at least one audio signal.

11. The active noise control system of any of claim 9 or 10, wherein said at least a part of said area is a secondary path, defined as the space between said loudspeaker and an error microphone (M_E) situated inside said local zone (CZ).

12. The active noise control system of claim 11, wherein said at least one physical measurement comprises an electrical impedance measured on said error microphone (M_E).

13. The active noise control system of any of claims 9 to 12, wherein said anti-noise signal is generated by a loudspeaker (LS).

14. The active noise control system of claim 13, wherein said at least one physical measurement comprises an indication of proximity of an object of said loudspeaker.

15. A computer program product comprising a computer readable medium, having thereon a computer program comprising program instructions, the computer program being loadable into a data-processing unit and adapted to cause execution of the method according to any of claims 1 to 8 when the computer program is run by the data-processing unit.

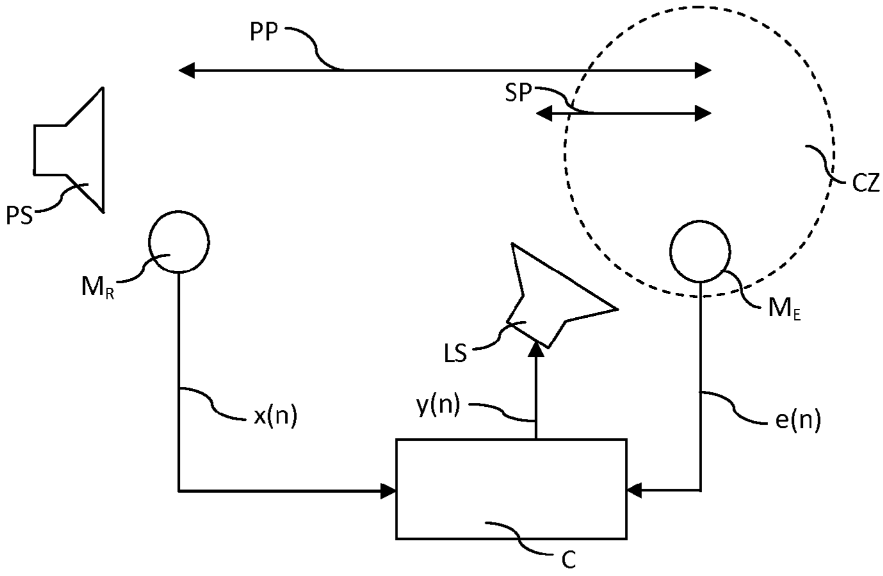
PP
SP
CZ
PS
M_R
LS
M_E
x(n)
y(n)
e(n)
C

Fig. 1

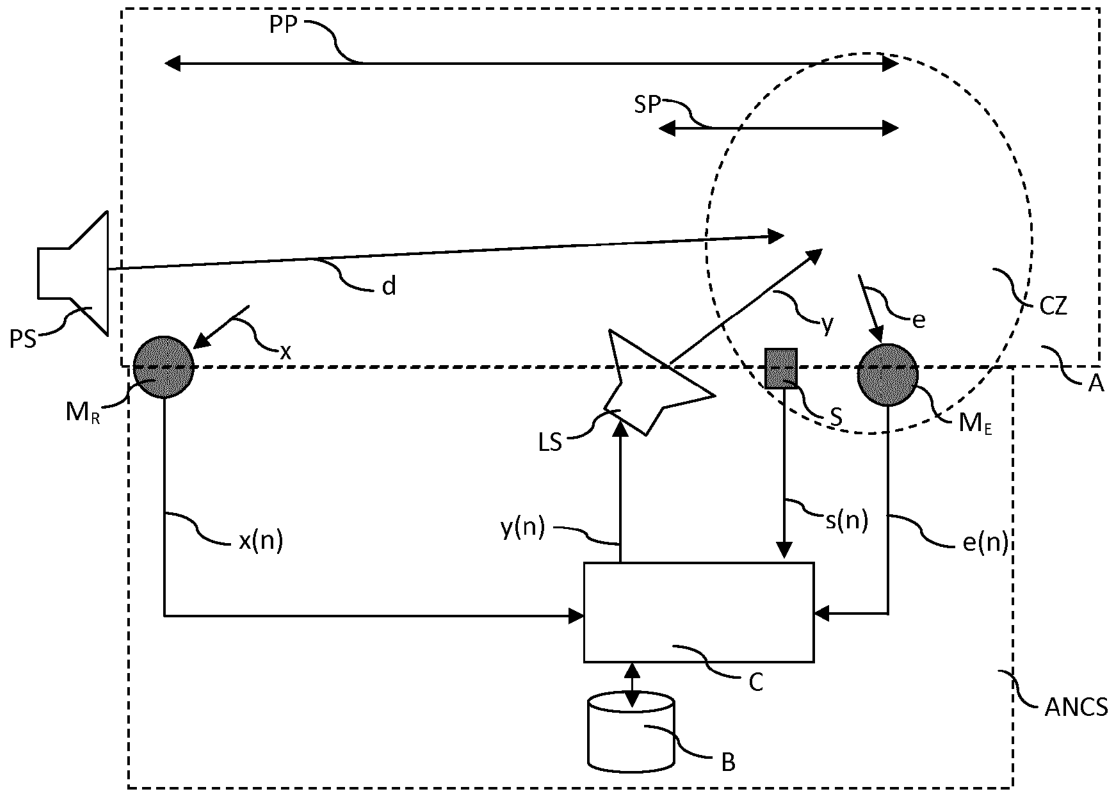
PP
SP
d
PS
x
y
e
CZ
A
M_R
LS
S
M_E
x(n)
y(n)
s(n)
e(n)
C
ANCS
B

Fig. 2

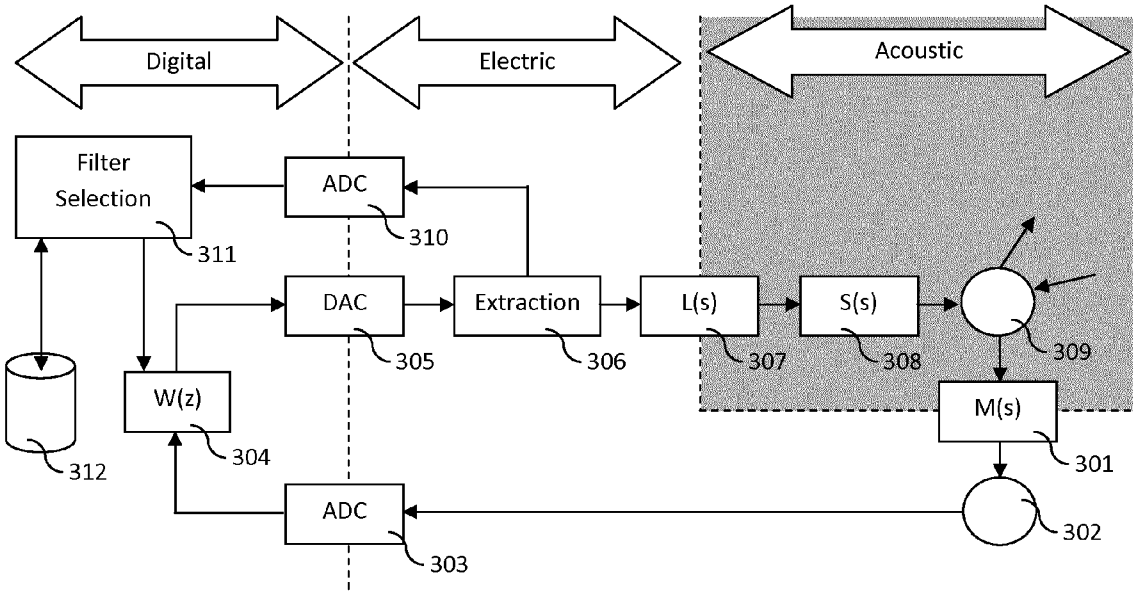
Digital
Electric
Acoustic
Filter Selection
ADC
310
311
DAC
Extraction
L(s)
S(s)
305
306
307
308
309
W(z)
304
312
M(s)
301
ADC
303
302

Fig. 3

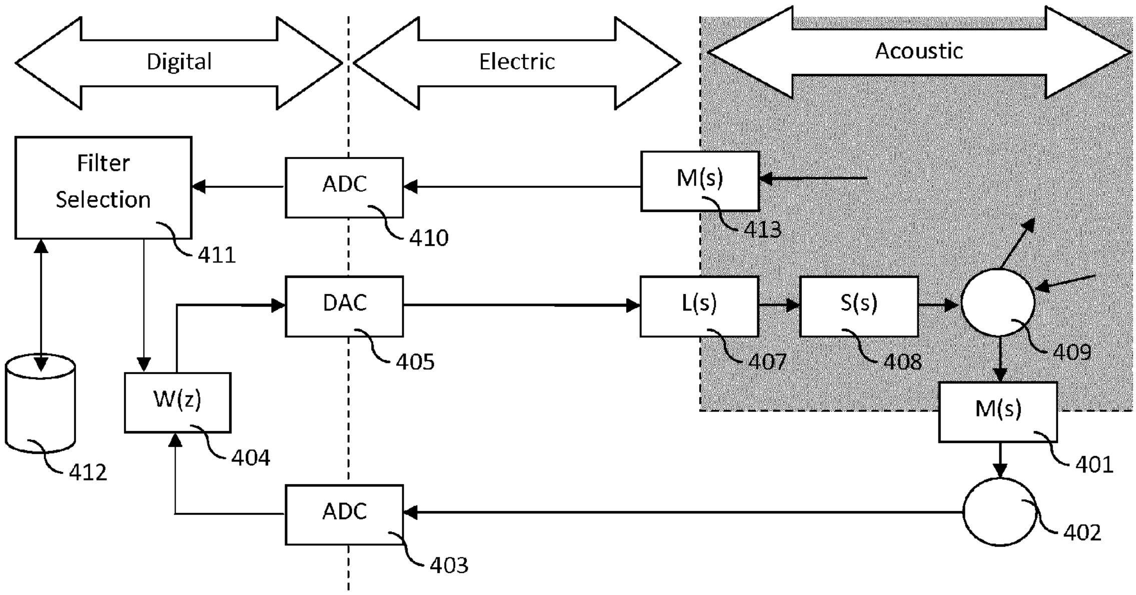
Digital
Electric
Acoustic
Filter Selection
ADC
M(s)
410
413
411
DAC
L(s)
S(s)
405
407
408
409
W(z)
404
412
M(s)
401
ADC
403
402

Fig. 4

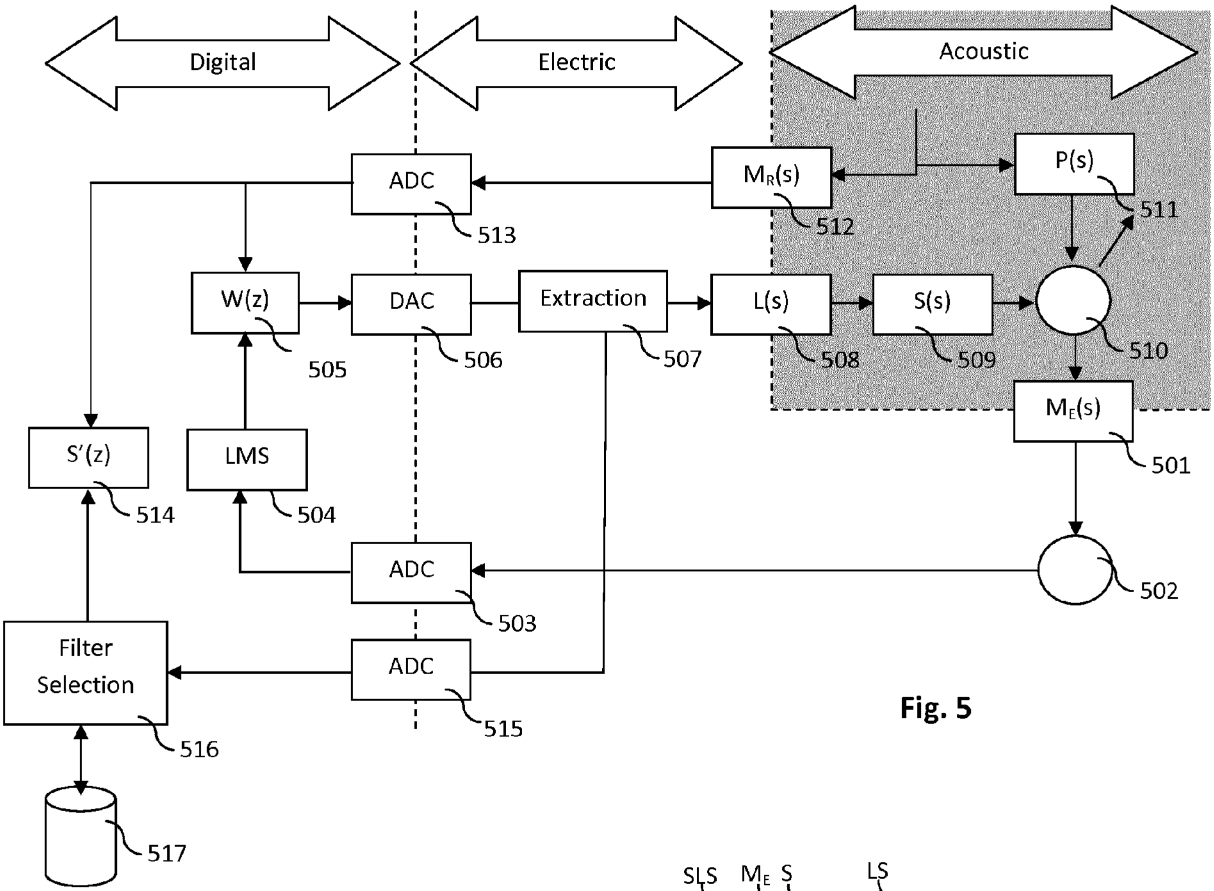
Digital
Electric
Acoustic
ADC
513
MR(s)
512
P(s)
511
W(z)
505
DAC
506
Extraction
507
L(s)
508
S(s)
509
510
ME(s)
501
S'(z)
514
LMS
504
ADC
503
502
Filter Selection
516
ADC
515
517

Fig. 5

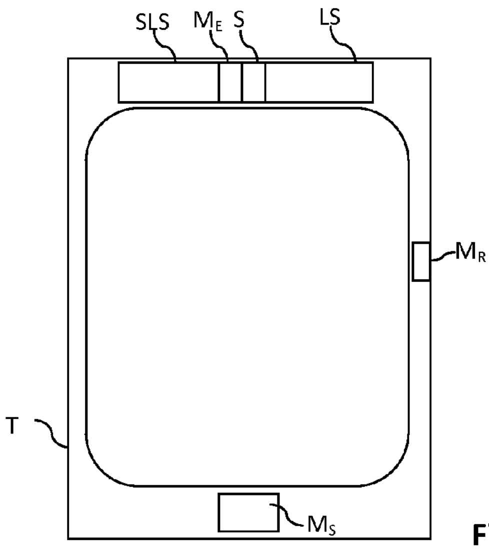
SLS
ME
S
LS
MR
T
MS

Fig. 6

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EUROPEAN SEARCH REPORT

Application Number
EP 12 30 6013

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (IPC)
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Y	* abstract; figures 2,5 * * paragraphs [0001], [0004] - [0007], [0026] - [0037] * -----	2,4,6, 10,12,14	
Y	US 2012/057718 A1 (VERNON SCOTT DENNIS [US]) 8 March 2012 (2012-03-08) * abstract; figure 2 * * paragraphs [0001], [0023], [0026], [0030] * -----	2,4,6, 10,12,14	
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TECHNICAL FIELDS SEARCHED (IPC)
G10K
H04R

The present search report has been drawn up for all claims

1

Place of search	Date of completion of the search	Examiner
The Hague	11 February 2013	Vollmer, Thorsten

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11-02-2013

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