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Description

TECHNICAL FIELD

[0001] The present invention relates to a method of, a device (and its use) and a system for enhancing the signal quality of an audio signal, e.g. in connection with the propagation of an audio signal to a listening device, e.g. a hearing aid. The invention further relates to a data processing system and to a computer readable medium.

[0002] The invention may e.g. be useful in applications such as listening devices, e.g. hearing aids, receiving audio sound from a signal source via an acoustic path.

BACKGROUND ART

[0003] In many wireless audio streaming scenarios the acoustical audio signal is present in parallel to a corresponding wireless electromagnetic signal, e.g. audio streaming from a TV, audio streaming in a class room, etc. The misalignment in time between the streamed audio signal and the acoustic audio signal is in many situations a problem. If the misalignment is more than 10 ms, sound quality begins to drop. If the misalignment is increased even more, audio-visual non-synchronicity begins to appear. If the delay is more than 50 ms, audio-visual non-synchronicity like e.g. lip-reading makes the situation quite unpleasant and decreases speech intelligibility.

[0004] WO 2008/071236 A2 describes a hearing system with enhanced noise cancelling to improve a S/N-ratio of a target signal S comprising noise N, the system comprising a selecting unit for selecting between an S+N-signal and an estimate N* of the noise signal and assigning one of the signals to be the signal comprising noise and the other signal to be the noise estimate, the system further comprising a filtering unit (e.g. an adaptive filter) for filtering the two input signals and providing an estimate S* of the target signal as an output. The S+N- and N*-signals may e.g. originate from a remote microphone, from a local microphone of a hearing aid, from a microphone of a contra-lateral hearing aid, etc.

[0005] EP 1 691 573 A2 describes a system with dynamic loudness adaptation comprising a remote microphone for picking up an audio signal and transmitting it to a hearing aid, the hearing aid comprising a local microphone for picking up environmental sounds, the hearing aid comprising a classification unit for determining a classification index based on the frequency and/or amplitude and/or temporal characteristics of the signals from the remote and local microphones and wherein a predefined amplitude and/or frequency ratio of the two signals is adapted based on the classification before the resulting signal is presented to a user of the hearing aid.

[0006] The present invention proposes among other things a solution to this problem.

DISCLOSURE OF INVENTION

[0007] The present invention deals in general with signal enhancement in listening systems. Embodiments of the invention relate to the handling of delay differences between acoustically propagated and wirelessly transmitted audio signals. Embodiments of the invention deal with the treatment of audio signals, which are to accompany video-images or real ('live') images of persons or scenes to be simultaneously perceived by a viewer. The idea is - in addition to the acoustically propagated audio signal - to wirelessly transmit (stream) the audio signal from an audio source, e.g. a TV-set or a wired or wireless microphone, to an audio receiver, e.g. a hearing aid.

[0008] In embodiments not according to the invention, the streamed audio signal is mainly used for building a signal model of the streamed signal source. This model is used to increase the signal-to-noise ratio of the acoustically propagated and received audio signal because the model can be used to determine which part of the input (signal+noise) is dominated by the signal, and which part is dominated by the noise.

[0009] In noise reduction algorithms, it is known to subtract an estimate of the noise from the mixed signal comprising signal and noise. In embodiments not according to the present invention, the 'opposite' is done in that the 'clean' version of the signal (the streamed audio signal) is used to extract characteristics of the target signal part of the received, acoustically propagated signal. Characteristics of the target signal include its frequency spectrum, periodicity, modulation at different frequencies f (e.g. modulation index, MI(f), top and bottom trackers, TT(f), BT(f), respectively) onset/offset characteristics, input level, etc. The extracted characteristics (the model) of the target signal can e.g. be used to adapt possible noise reduction and compression algorithms to provide the same characteristics in the processed version of the received acoustically propagated signal. Such processing can e.g. be performed in a signal processing unit of a listening device.

[0010] The present scheme can further be used e.g. to filter out noise from distinct sources, e.g. a ventilator, a household appliance or the like using a directional microphone system or, alternatively, if the noise has its origin from in front of the person wearing the hearing aid, to reduce it using a noise reduction algorithm.

[0011] Similarly, the concept can be used in connection with 'own voice detection' by using a specific 'own voice detector' to extract characteristics of the 'own voice' and wirelessly transmit those characteristics (or alternatively the full audio signal comprising 'own voice') to a hearing aid of another listening person, which can then be specifically 'tuned' to the reception of that particular voice.

[0012] The concept can be used to add spatial information about the present location of a user (e.g. a particular room) to a wirelessly streamed audio signal with the purpose of adding directional information, etc., to the otherwise 'clean' streamed signal.

[0013] An object of embodiments of the present invention is to provide a scheme for improving signal quality of an audio signal received by a listening device.

[0014] Objects of the invention are achieved by the invention described in the accompanying claims and as described in the following.

A method of enhancing an audio signal:

[0015]

- An object of the invention is achieved by a method of enhancing an audio signal in a receiving device as defined in claim 1.

[0016] An advantage of the invention is that a target signal is enhanced.

[0017] Another advantage of embodiments of the invention is that the acoustically propagated signal is enhanced without introducing a further delay in its propagation path.

[0018] Another advantage of embodiments of the invention is that the streamed signal can be used to precisely estimate the impulse response of the path from the loud speaker generating the (acoustic version of the) audio signal to the microphone of the listening device, e.g. a hearing aid (i.e. dependent of the room in which the user is located). This estimate can then be more precisely de-convolved in the listening device (than if the source signal is unknown).

[0019] In the present context, the term 'streaming', refers to the transmission and reception of a (typically digital, e.g. encoded) signal, typically representing audio or video data, which is continuously generated (or transmitted from a stored file) and presented to a user or used in a medium as it is received. Typically, the streamed signal is presented to a user as it is received, without being permanently stored (apart from necessary buffering).

[0020] In the present context, the term an adaptive system, refers to a system that is able to respond to changes in its inputs. An adaptive system typically comprises a feedback loop. An example of an adaptive system is an adaptive filter comprising a variable filter part and an update algorithm part, the variable filter part providing a transfer function that is automatically adjusted to changing inputs based on an optimizing algorithm of the update algorithm part.

[0021] In an embodiment, the receiving device is adapted to be able to perform signal processing in separate frequency ranges or bands.

[0022] In an embodiment, the input side of the forward path of the receiving device comprises an AD-conversion unit for sampling an *analogue* electric input signal with a sampling frequency f_s and providing as an output a *digitized* electric input signal comprising digital time samples S_n of the input signal (amplitude) at

consecutive points in time $t_n=n*(1/f_s)$, where n is an integer. The duration in time of a sample is thus given by $T_s=1/f_s$. In general, the sampling frequency is adapted to the application (available bandwidth, power consumption, frequency content of input signal, necessary accuracy, etc.). In an embodiment, the sampling frequency f_s is in the range from 8 kHz to 40 kHz, e.g. around 16 kHz.

[0023] In an embodiment, the receiving device comprises a TF-conversion unit for providing a time-frequency representation of a signal. In an embodiment, the time-frequency representation comprises an array or map of corresponding complex or real values of the signal in question in a particular time and frequency range. In an embodiment, the TF conversion unit comprises a filter bank for filtering a (time varying) input signal and providing a number of (time varying) output signals each comprising a distinct frequency range of the input signal. In an embodiment, the TF conversion unit comprises a Fourier transformation unit for converting a time variant input signal to a (time variant) signal in the frequency domain. In an embodiment, the frequency range considered by the receiving device from a minimum frequency f_{min} to a maximum frequency f_{max} comprises a part of the typical human audible frequency range from 20 Hz to 20 kHz, e.g. from 20 Hz to 12 kHz. In an embodiment,-the frequency range f_{min}-f_{max} considered by the receiving device is split into a number P of frequency bands, where P is e.g. larger than 5, such as larger than 10, such as larger than 50, such as larger than 100, at least some of which are processed individually.

[0024] In a particular embodiment, the method comprises estimating the delay difference between the *propagated* electric signal and the *streamed* target audio signal or signals originating there from. Which one of the signals that arrives first in the receiving device in (electrical form) will depend on the physical length of the acoustic propagation path and the latency of the wireless link, e.g. on delays in transceivers of the wireless transmission path (including in possible coding-decoding units, modulation-demodulation units, etc.) for transmitting and receiving the electromagnetic signal, and on delays in the input transducer, possible front-end amplifiers and/or other processing of the acoustically propagated signal during reception, etc. In some applications, the (acoustically) *propagated* electric signal will have the lowest delay. This *may* e.g. the case, if the wireless link is based on inductive coupling between transmitter and receiver. In other cases the (electromagnetically) *streamed* target audio signal will have the lowest delay. This *may* e.g. be the case, if the wireless link is based on radiated fields.

[0025] In a particular embodiment, the method comprises using the resulting delay difference in the estimation of the target signal.

[0026] In an aspect not according to the invention, the idea is to use the streamed audio signal ONLY for building a signal model of the streamed signal source. This model is used in a signal enhancement system like e.g. the "Spectral Subtraction" algorithm (see e.g. [Boll, 1979]). This type of algorithm uses an estimate of the noise and by comparing this estimate with the input (signal+noise) the optimal gain is calculated. A perfect estimate of the *signal* is available (the streamed target audio signal) and by comparing this to the input (signal+noise) we can calculate an

optimal gain (we can call this a reversed Spectral Subtraction or a Spectral Enhancement algorithm). Alternatively, a Wiener filter could be used (cf. e.g. [Widrow et al., 1975]).

[0027] Some algorithms use the signal estimate directly, e.g. the formant tracking algorithms like HMM (Hidden Markov Model) (see e.g. [Rabiner, 1989]) or Linear Prediction methods (see e.g. [Makhoul, 1975]). In this case, the streamed signal is used to extract signal model information like formants, spectral shape, etc., and use this in the enhancement algorithm. According to an aspect not according to the present invention, the streamed signal is NOT used in the direct signal path (the streamed signal is not presented to a user, cf. e.g. embodiments of FIG. 2b, 2c, 2g, 3a, 3b, 5. 7a, 8, 12a). It is mainly used to extract information about the acoustically propagated target signal (such information (the model) being passed to the Signal Enhancement algorithm for estimating (enhancing) the target signal). In this way we do not have any problems with the link delay because the model can be updated quite slowly and we can expect a significant increase in the signal-to-noise ratio.

[0028] In embodiments of this aspect, the method comprises estimating the target signal *from the propagated* electric signal using the streamed target audio signal or a signal derived there from as an input to the adaptive algorithm to improve the estimate of the target signal. Here, the (possibly delayed) *propagated* electric signal is e.g. fed to the variable filter part of an adaptive filter whereas the (possibly delayed) *streamed* target audio signal is used in the algorithm part of the adaptive filter to update filter coefficients of the variable filter part. This has the advantage of increasing the signal to noise ratio of the propagated electric signal.

[0029] In an aspect of the present invention, the acoustically propagated signal is used to add *spatial* information about the present location of a user (e.g. a particular room) to a wirelessly streamed audio signal with the purpose of adding directional information, etc., to the otherwise 'clean' streamed signal (the resulting 'enhanced' streamed signal being presented to a user, cf. e.g. embodiments of FIG. 2d, 2e, 6, 7b).

[0030] In embodiments of this aspect of the invention, the method comprises estimating the target signal *from the streamed* target audio signal using the propagated electric signal or a signal derived there from as an input to the adaptive algorithm to improve the estimate of the target signal. This can e.g. be implemented by an adaptive filter by feeding the (possibly delayed) *streamed* target audio signal to the variable filter part of an adaptive filter whereas the (possibly delayed), *propagated* electric signal is used in the algorithm part of the adaptive filter to update filter coefficients of the variable filter part.

[0031] In a particular embodiment, the method comprises

- delaying the relevant one of the signals, the *estimate* of the target signal, the *propagated* electric signal and the *streamed* target audio signal, or signals originating there from with the estimated delay difference; and

- using the resulting signal in the estimation of the target signal.

[0032] In a particular embodiment, the method comprises extracting characteristics of the target signal from the streamed target audio signal. In an embodiment, the method additionally comprises extracting characteristics of the target signal from the propagated electric signal. In an embodiment, the characteristics of the target signal include one or more of the following: the frequency spectrum, modulation at different frequencies (e.g. modulation index, e.g. top and bottom trackers of a modulation index vs. frequency), onset/offset characteristics, input level, etc. In an embodiment, the method comprises comparing corresponding characteristics (e.g. modulation index or input level) of extracted from the streamed target audio signal and the propagated electric signal, respectively.

[0033] In a particular embodiment, the method comprises using the extracted characteristics of the streamed target audio signal as inputs to processing algorithms for improving the estimated target signal. In an embodiment, the method comprises using a comparison of corresponding characteristics (e.g. modulation index or input level) extracted from the streamed target audio signal and the propagated electric signal, respectively as inputs to processing algorithms for improving the estimated target signal. In an embodiment, algorithms comprise one or more algorithms for processing of gain, directionality, noise reduction or compression, etc., to appropriately adapt (enhance) characteristics of the target signal estimate.

[0034] In an embodiment, the estimated target signal is further improved in further processing algorithms, e.g. by adapting the estimated target signal according to a user's needs.

[0035] In a particular embodiment, the extracted characteristics of the streamed target audio signal or a *comparison* between characteristics of the streamed target audio signal and the propagated electric signal are used to compensate for non-linearities in loudspeakers in a room, thereby an improved sound quality can be provided, while maintaining other sounds from the environment. This has the advantage that the resulting estimated version of the target sound signal is NOT 'destroyed' by bad components in loudspeaker(s) providing the sound source target signal.

[0036] In a particular embodiment, the extracted characteristics of the streamed target audio signal or a *comparison* between characteristics of the streamed target audio signal and the propagated electric signal are used to remove noise from distinct audio sources in the environment of the receiving device, e.g. from a household appliance, e.g. a dish washing machine, a ventilator, etc.

[0037] In a particular embodiment, the method comprises extracting characteristics of the acoustic propagation path from the propagated acoustic signal.

[0038] In a particular embodiment, the characteristics of the acoustic propagation path include one or more of the following: interaural difference cues, distance information, intensity, direct to reverberant energy ratio, room impression.

[0039] In a particular embodiment, the extracted characteristics of the acoustic propagation path are used to add spatial information to the target signal estimate, e.g. characteristics of the room, reflections, background sounds, directional cues, reverberation, etc.

[0040] In a particular-embodiment, the propagated acoustic signal is attenuated, e.g. cancelled, in or by the receiving device before being presented to a user, e.g. in hearing aids or headphones to be able to fully control the sound presented to a user.

[0041] In an embodiment, the method is used in a listening device, e.g. a protective device, a head phone or a headset, a hearing aid or a pair of hearing aids of a binaural fitting. An advantage of embodiments of the invention is that the delay problem is solved, and further that the user gets the audio signal through their own ears (via the hearing aid, i.e.) including additional background sounds, so that an experience of being cut-off from the environment is avoided. A further advantage of embodiments of the invention is that the target signal is enhanced compared to the acoustically of wirelessly propagated signals comprising the target signal.

Audio enhancement device:

[0042] An audio enhancement device for enhancing an audio signal is furthermore provided by the present invention as defined in claim 11.

[0043] It is intended that the process features of the method described above, in the detailed description of 'mode(s) for carrying out the invention' and in the claims can be combined with the (audio enhancement) device, when appropriately substituted by a corresponding structural feature and vice versa. Embodiments of the device have the same advantages as the corresponding method.

[0044] In a particular embodiment, the audio enhancement device comprises a first estimator unit for estimating the delay difference between the *propagated* electric signal and the *streamed* target audio signal or signals originating there from. In a particular embodiment, the audio enhancement device is adapted for using the resulting delay difference in the estimation of the target signal.

[0045] In a particular embodiment not according to the invention, the first adaptive system is adapted to base its estimate of the target signal on the *propagated* electric input signal and said estimated delay difference.

[0046] In a particular embodiment, the first adaptive system is adapted to base its

estimate of the target signal on the *streamed* target audio signal and said estimated delay difference.

[0047] In a particular embodiment, the audio enhancement device comprises a second estimator unit for estimating characteristics of the target signal from the streamed target audio signal. In an embodiment, the characteristics of the target signal include one or more of the following: the frequency spectrum, modulation at different frequencies (e.g. modulation index, e.g. top and bottom trackers of a modulation index vs. frequency), onset/offset characteristics, etc. In a particular embodiment, the audio enhancement device is adapted to provide that the extracted characteristics of the streamed target audio signal are used to as inputs to processing algorithms for improving the target signal. In an embodiment, the estimated target signal is thereby improved (e.g. according to a user's needs), e.g. by adapting algorithms for gain, directionality, noise reduction or compression, etc., to provide the same characteristics in the processed version of the target signal estimate as in the streamed target audio signal.

[0048] In a particular embodiment, the audio enhancement device comprises a third estimator unit for estimating characteristics of the acoustic propagation path from the propagated acoustic signal. In a particular embodiment, the characteristics of the acoustic propagation path include one or more of the following: interaural difference cues, distance information, intensity, direct to reverberant energy ratio, room impression. In a particular embodiment, the audio enhancement device is adapted to provide that the extracted characteristics of the acoustic propagation path are used to add spatial information to the target signal estimate. In a particular embodiment, the spatial information comprises e.g. characteristics of the room, reflections, background sounds, directional cues, reverberation, etc.

[0049] In a particular embodiment, said first adaptive system comprises an adaptive filter for providing said estimate of the target signal, the adaptive filter comprising an algorithm part and a variable filter part where the algorithm part is adapted to update a filter characteristic of the variable filter part.

[0050] In a particular embodiment, the first estimator unit comprises an adaptive filter for providing said estimate of the delay difference.

[0051] In a particular embodiment, the audio enhancement device comprises a signal processing unit for further processing said estimate of the target signal, e.g. for running processing algorithms for improving the target signal and/or for adding spatial information to the estimate of the target signal. The signal processing unit may be adapted to further process the estimate of the target signal according to a user's needs.

[0052] In a particular embodiment, the audio enhancement device comprises an output transducer for presenting the estimate of the target signal or an output from said signal processing unit comprising a further processing of said estimate of the target signal to a user. In an embodiment, the audio enhancement device

comprises an output transducer for presenting the estimate of the target signal or an output from said signal processing unit comprising a further processing of said estimate of the target signal as a stimulus adapted to be perceived by a user as an output sound (e.g. an output transducer (such as a number of electrodes) of a cochlear implant or of a bone conducting hearing device.

[0053] In an embodiment, the audio enhancement device form part of a listening device, e.g. a hearing instrument, a head set, a head phone, or an ear protection device, or a combination thereof.

Audio enhancement system:

[0054] An audio enhancement system is moreover provided by the present invention. The audio enhancement system comprises an audio source for generating an acoustic target signal and a transmitting device for generating a wireless signal comprising a clean representation of said target signal in the form of a target audio signal and a receiving device comprising an audio enhancement device as described above, in the detailed description of 'mode(s) for carrying out the invention' and in the claims.

[0055] In a particular embodiment, the *transmitting device* is embodied in an entertainment device comprising a microphone and/or produce images and accompanying sound signals. The transmitting device, can e.g. be an A/V device (A/V=Audio/Visual), e.g. a TV-set, a PC, a wired or wireless microphone, a karaoke system, etc. In an embodiment, the *entertainment device* comprises a loudspeaker for propagating a target sound, a wireless transmitter for electromagnetically propagating the sound and a microphone for picking up a target sound (or a part thereof) from a speaker or singer or another intended sound from the environment. In an embodiment, such device comprising a microphone comprises a PC and/or a karaoke-device.

[0056] In a particular embodiment, the *receiving device* is embodied in a listening device, e.g. a body-worn listening device, e.g. comprising a headphone, a head set, an ear protection device and/or a hearing instrument.

[0057] In an embodiment, the *audio source* comprises a loudspeaker.

[0058] In an embodiment, the *audio source* is embodied in an entertainment device comprising images and accompanying sound signals (such as an A/V device, e.g. a TV-set or a PC).

[0059] In an embodiment, the *audio source* and said *transmitting device* are integrated in one physical device comprising a common housing.

[0060] In an embodiment, the *audio source* is a voice, e.g. a voice of a human being.

[0061] In an embodiment, the *transmitting device* comprises a microphone or a

listening device adapted for being worn by a user and comprising an 'own voice detector' for detecting and extracting the users own voice or characteristics thereof, the audio source being the user's own voice, and the transmitting device being adapted to wirelessly transmit an audio signal comprising the user's own voice to said receiving device. This has the advantage that the receiving audio enhancement device, e.g. a hearing aid, of another listening person, which can be specifically 'tuned' to the reception of the voice of the wearer of the transmitting listening device, e.g. a microphone or a hearing aid.

[0062] In an embodiment, the *transmitting device* is an audio enhancement device as described above, in the detailed description of 'mode(s) for carrying out the invention' and in the claims.

Use:

[0063] Use of an audio enhancement device or of an audio enhancement system as described above, in the detailed description of 'mode(s) for carrying out the invention' and in the claims is moreover provided by the present invention. In an embodiment, use of an audio enhancement device in a device selected from the group of listening devices comprising a headset, an active earplug, a headphone, a hearing instrument and combinations thereof is provided. In an embodiment, use of an audio enhancement system in a public address system or a karaoke system is provided.

A tangible computer-readable medium:

[0064] A tangible computer-readable medium storing a computer program comprising program code means for causing a data processing system to perform at least some of the steps of the method described above, in the detailed description of 'mode(s) for carrying out the invention' and in the claims, when said computer program is executed on the data processing system is furthermore provided by the present invention. In addition to being stored on a tangible medium such as diskette-, CD-ROM-, DVD-, or hard disk-media, or any other machine readable medium, the computer program can be transmitted via a transmission medium such as a wired or wireless link or a network, e.g. the Internet, and loaded into a data processing system for being executed at a location different from that of the tangible medium.

A data processing system:

[0065] A data processing system comprising a processor and program code means for causing the processor to perform at least some of the steps of the method described above, in the detailed description of 'mode(s) for carrying out the invention' and in the claims is furthermore provided by the present invention.

[0066] Further objects of the invention are achieved by the embodiments defined in the dependent claims and in the detailed description of the invention.

[0067] As used herein, the singular forms "a," "an," and "the" are intended to

include the plural forms as well (i.e. to have the meaning "at least one"), unless expressly stated otherwise. It will be further understood that the terms "includes," "comprises," "including," and/or "comprising," when used in this specification, specify the presence of stated features, integers, steps, operations, elements, and/or components, but do not preclude the presence or addition of one or more other features, integers, steps, operations, elements, components, and/or groups thereof. It will be understood that when an element is referred to as being "connected" or "coupled" to another element, it can be directly connected or coupled to the other element or intervening elements maybe present, unless expressly stated otherwise. Furthermore, "connected" or "coupled" as used herein may include wirelessly connected or coupled. As used herein, the term "and/or" includes any and all combinations of one or more of the associated listed items. The steps of any method disclosed herein do not have to be performed in the exact order disclosed, unless expressly stated otherwise.

BRIEF DESCRIPTION OF DRAWINGS

[0068] The invention will be explained more fully below in connection with a preferred embodiment and with reference to the drawings in which:

FIG. 1 shows embodiments of an audio enhancement system according to the disclosure comprising an audio source, a transmitter and one or more listening devices comprising an audio enhancement device according to the disclosure;

FIG. 2 shows block diagrams of various audio enhancement devices according to embodiments of the disclosure;

FIG. 3 shows block diagrams of two embodiments of an audio enhancement device according to the disclosure, FIG. 3a showing a single-microphone device and FIG. 3b showing a multi-microphone device;

FIG. 4 shows a listening device comprising an audio enhancement device according to an embodiment of the disclosure;

FIG. 5 shows an example of an audio enhancement system according to an embodiment of the disclosure, the system comprising an audio enhancement device wherein the target signal is estimated based on the acoustically propagated signal based on LMS deconvolution;

FIG. 6 shows an audio enhancement system according to an embodiment of the disclosure, the system comprising an audio enhancement device wherein the target signal is estimated based on the electromagnetically propagated signal;

FIG. 7 shows block diagrams of two embodiments of a listening device according to the disclosure;

FIG. 8 shows an audio enhancement system adapted for enhancement of a particular voice according to an embodiment of the disclosure;

FIG. 9 shows a flow diagram of a method according to an embodiment of the present disclosure;

FIG. 10 shows an example of characteristics of the streamed target audio signal, here modulation index MI vs. time t (FIG. 10a) and resulting inputs to a processing algorithm providing gain G [dB] vs. modulation index MI;

FIG. 11 shows an example of a comparison of characteristics of the streamed target audio signal with those of the acoustically propagated signal, here top and bottom trackers of modulation index MI vs. time t (FIG. 11 a) and resulting inputs to processing algorithms, here providing incremental gain ΔG [dB] vs. frequency f (FIG. 11 b) and incremental noise reduction ΔNR [dB] vs. frequency f (FIG. 11 c); and

FIG. 12 shows an audio enhancement device according to an embodiment of the disclosure (FIG. 12A) and corresponding characteristics of the target signal, here level differences ΔG [dB] vs. frequency f (FIG. 12b).

[0069] The figures are schematic and simplified for clarity, and they just show details which are essential to the understanding of the invention, while other details are left out. Throughout, the same reference numerals or names are used for identical or corresponding parts.

[0070] Further scope of applicability of the present invention will become apparent from the detailed description given hereinafter. However, it should be understood that the detailed description and specific examples, while indicating preferred embodiments of the invention, are given by way of illustration only, since various changes and modifications within the spirit and scope of the invention will become apparent to those skilled in the art from this detailed description.

MODE(S) FOR CARRYING OUT THE INVENTION

[0071] FIG. 1 shows embodiments of an audio enhancement system according to the invention comprising an audio source, a transmitter and one or more listening devices comprising an audio enhancement device according to the invention.

[0072] The audio enhancement system of FIG. 1 a comprises a TV-set and a pair of listening devices, here a pair of hearing instruments of a binaural fitting. The TV-set 1 is provided with a loudspeaker for acoustically emitting a sound signal ***(APTS)*** 6 (a target signal, which a user wishes to receive) corresponding to the TV-images AND with a transmitter for transmitting the same sound (termed target audio signal) via a wireless link 4 in the form of signal ***WLS***. In addition to the target signal 6, a noise signal **(*N*)** 8 (here produced by a fan 7, but representing all background noise (including other sound sources than the target sound) in the environment of the user) is mixed with the acoustically propagated target signal. Both signals are received by the pair of hearing instruments (HI) 3, each comprising an input transducer for converting the acoustically propagated sound signal(s) 6, 8 **(*APTS+N*)** to an electric input signal in the respective HIs, AND at least one of the HIs comprising a receiver for receiving the wirelessly transmitted

signal (***WLS***) 4 and extracting a streamed target audio signal (which is typically not in phase with the acoustically propagated target signal). The wireless link can be based on a near-field coupling, e.g. an inductive coupling, between inductors of the TV-transmitter and the HI receiver(s) or based on radiated (far-field) electromagnetic fields. The transmission can be based on analogue or digitally modulated signals. In the embodiment of FIG. 1a, a link based on radiated fields and operating according to the Bluetooth specification is anticipated (cf. transmitter *BT-Tx* of the TV-set and receiver *BT-Rx* of the hearing instrument(s)). A communication between the two hearing instruments allowing the exchange of control and/or status information and/or audio signals is preferably implemented. This wireless link can e.g. be based on near-field or far-field electromagnetic communication. Various aspects of inductive communication are discussed e.g. in EP 1 107 472 A2 and US 2005/0110700 A1. WO 2005/055654 and WO 2005/053179 describe various aspects of a hearing aid comprising an induction coil for inductive communication with other units. US 2008/0013763 A1 describes a system for wireless audio transmission with a low delay from a transmission device (e.g. a TV-set) to a hearing device, e.g. based on radiated fields and Bluetooth. A wireless link protocol is e.g. described in US 2005/0255843 A1.

[0073] FIG. 1b shows an embodiment of an audio enhancement system comprising a wireless microphone ***M*** located at a variable position MP(t) = [$X_m(t)$, $Y_m(t)$, $Z_m(t)$] (t being time, and X, Y, Z being the coordinates of the position in an xyz-coordinate system) for picking up the voice (mixed with possible noise in the environment of the microphone) of a speaker ***S*** located at a variable position SP(t) = [$X_s(t)$, $Y_s(t)$, $Z_s(t)$], the wireless microphone being adapted to wirelessly transmit the picked up target signal. The system may further comprise a broadcast access point ***BAP*** located at a fixed position BP = [X_{bp}, Y_{bp}, Z_{bp}], and adapted for relaying the radio signal from the wireless microphone. The system additionally comprises a pair of listening devices (e.g. hearing aids) worn at the ears of a listener ***L*** located at a variable position LP(t) = [$X_l(t)$, $Y_l(t)$, $Z_l(t)$] and adapted to receive the wirelessly transmitted (audio) signal from the wireless microphone (e.g. via the broadcast access point) as well as the directly propagated audio signal from the speaker (mixed with possible other sounds and acoustic noise from the surroundings of the user). $A_R(f,t)$, $A_L(f,t)$, $A_{mic}(f,t)$ represent acoustic transfer functions *from the speaker* to the *R*ight hearing instrument, to the *L*eft hearing instrument and to the wireless *mic*rophone, respectively. The acoustic transfer functions A(f,t) are dependent on frequency f and time t. The acoustic propagation delay in air is around 3 ms/m (i.e. propagation path of 10 m's length induces a delay of around 30 ms in the acoustically propagated signal). $R_T(f)$ and $R_F(f)$ represent radio transfer functions from the wireless microphone to the broadcast access point and from the broadcast access point to the hearing instruments, respectively (assumed equal for the two left and right HI-positions). The radio transfer functions R(f) are dependent on frequency f but assumed independent of time.

[0074] FIG. 1c illustrates an application of an audio enhancement system in a small group comprising a speaker ***S*** speaking a target signal into a microphone ***M*** (here a wireless microphone) comprising a transmitter ***Tx*** for wirelessly

transmitting a signal ***WLS*** comprising the electrical target audio signal picked up by the microphone to one or more listeners ***L*** (here three are shown) wearing a listening device ***LD*** at one or both ears. The spoken target signal ***ATS*** is acoustically propagated to the listeners (and typically distorted, attenuated and mixed with other sounds along the path to the listener(s), as indicated by signals ***APS, APS'***). The listening device ***LD*** comprises an audio enhancement device for estimating the target signal from the acoustically *propagated* electric signal and the *streamed* target audio signal using an adaptive system.

[0075] FIG. 1d illustrates an application of an audio enhancement system in a public address system, e.g. in a class room or auditorium or entertainment application (e.g. karaoke), where a speaker ***S*** (e.g. a teacher) speaks or sings a target signal (*myyyyy waaaayy*) into a microphone ***M*** (e.g. a wireless microphone, here a wired microphone is shown) in communication with a base station ***BS*** comprising circuitry for driving a loudspeaker (and possibly adding music or other sounds to the signal) for acoustically propagating the resulting signal to one or more listeners ***L*** (the signal being typically distorted, attenuated and mixed with other sounds along the path to the listener(s) as indicated by the diminishing character size in *myyyyy waaaayy).* The base station further comprises a transmitter ***Tx*** for wirelessly transmitting a signal ***WLS*** comprising the electrical target audio signal picked up by the microphone (and possibly added sound, e.g. accompanying music) to one or more listeners ***L*** wearing a listening device at one or both ears (here three are shown wearing listening devices ***LD*** and two listeners are shown only receiving the acoustically propagated signal). The listening devices comprise an -audio enhancement device for estimating the target signal from the acoustically *propagated* electric signal and the *streamed* target audio signal using an adaptive system.

[0076] FIG. 2 shows block diagrams of various audio enhancement devices according to embodiments of the disclosure.

[0077] The embodiments of FIG. 2a, 2b, 2c, 2d, 2e, 2f, 2g, 2h and 2i each comprise a receiver of a wireless signal comprising an antenna, an amplifier and a demodulator adapted for extracting a target audio signal (termed the *streamed* target audio signal) from the wirelessly received signal. The receiver of a wireless signal may additionally comprise analogue to digital (AD) converter and/or time to frequency (t->f) conversion units (cf. e.g. filter bank unit *FB* in FIG. 2h, 2i). The audio enhancement device further comprises at least one microphone (in the embodiments of FIG. 2a, 2b, 2c, 2d, 2e and 2f one microphone is indicated, whereas the embodiment in FIG. 2g comprises a multitude of microphones m_1, ..., m_n) for converting an input sound to an electrical input signal, the input sound e.g. comprising a mixture of a target signal from an acoustic source and noise signals from the environment (termed the *propagated* electric signal). The microphone(s) may additionally comprise analogue to digital converting units and/or a time to frequency conversion units (or such units may be implemented elsewhere in the audio enhancement device depending on practical circumstances, cf. e.g. filter banks *FB* in FIG. 2h, 2i).

[0078] In the most general embodiment of FIG 2a, the audio enhancement device (*AE* in FIG. 2a) - in a addition to the receiver circuitry for receiving the acoustically and wirelessly propagated signals - further comprises an audio enhancement unit (AS in FIG. 2a) receiving as inputs the *streamed* target audio signal comprising the target audio signal and the acoustically propagated signal, the audio enhancement unit comprising an adaptive system (e.g. an adaptive filter) adapted for estimating the target signal from the *propagated* electric signal and the *streamed* target audio signal and providing an enhanced target signal as an output *(OUT* in FIG. 2a). The output can e.g. be further processed in a signal processing unit (e.g. introducing user specific processing to adapt the signal to a particular user's hearing needs).

[0079] In the embodiments of FIG. 2b, 2c, 2d, 2e and 2g, the audio enhancement unit *(AS* in FIG. 2a) comprises an estimator unit (*E* in FIG. 2b, 2c, 2d, 2e and *ESTIMATOR* in FIG. 2g) and an actuator unit *(A* in FIG. 2b, 2c, 2d, 2e and *ACTUATOR* in FIG. 2g), one unit taking the *propagated* electric signal as a first input, the other taking the *streamed* target audio signal as a first input. The estimator unit is adapted for extracting characteristics of the input signal(s) and providing a control output comprising such characteristics as a second input to the actuator unit. The actuator unit is adapted for providing as an output *(OUT)* an estimate of the target signal based on its first and second inputs. In the embodiments of FIG. 2b, 2c, 2d, 2e and 2g, the enhanced target signal as an output *(OUT)* of the actuator unit is fed to the estimator unit and used in the extraction of said characteristics.

[0080] FIG. 2b and 2c illustrates embodiments, where the target signal is estimated based on the acoustically propagated signal (*propagated* electric signal) using the *streamed* target audio signal to extract characteristic of the target signal. In the embodiment of FIG. 2c, the *propagated* electric input signal to the actuator unit is additionally fed to the estimator unit and used in the extraction of the characteristics of the target signal.

[0081] FIG. 2d and 2e illustrates embodiments where the target signal is estimated based on the electromagnetically propagated signal (the *streamed* target audio signal) using the acoustically *propagated* electric signal to extract characteristics of the propagation path (room characteristics, distance, head related transfer functions, etc.). In the embodiment of FIG. 2e, the *streamed* target audio input signal to the actuator unit is additionally fed to the estimator unit and used in the extraction of the characteristics of the target signal (e.g. concerning the influence of the room).

[0082] FIG. 2f shows an embodiment of an audio enhancement device as in FIG. 2a, additionally comprising a control output signal CTR2 comprising characteristics of the target signal extracted from the *streamed* target audio input signal and/or from the *propagated* electric signal e.g. for use as an input to other processing algorithms further down the forward path of the estimated target signal, e.g. for adapting the target signal to a user's hearing profile e.g. of a communication

device, such as a mobile telephone or a listening device, such as a hearing instrument.

[0083] FIG. 2g shows an embodiment of an audio enhancement device comprising a multitude of microphones where the target signal is estimated based on the acoustically propagated signal (*propagated* electric signal). The audio enhancement device comprises an *ESTIMATOR* unit for extracting characteristics (a model) of a target signal from the wirelessly received target audio signal *WIN.* Additionally, the electrical input signal(s) *AIN1, AIN2, ..., AINn* from the microphones m_1, ..., m_n, and the estimate of the target signal *OUT* are fed to the *ESTIMATOR* unit as inputs. The *ESTIMATOR* unit provides as an output a control signal *CTRL* indicative of characteristics of the target signal. The audio enhancement device further comprises an *ACTUATOR* unit for adapting the electrical input signals from microphones m_1, ..., m_n to provide as an output *OUT* an estimate of the target signal based on the electrical input signals from microphones and the control signal(s) *CTRL* from the *ESTIMATOR* unit. The output *OUT* is fed back to the *ESTIMATOR* unit and used in the determination of the control signal output *CTRL.* The output *OUT* of the *ACTAUTOR* unit is e.g. used as an input to a signal processing unit for further signal processing relevant for the device in question which the audio enhancement device forms part of, e.g. a listening device, such as a hearing aid. Further signal processing may e.g. include adapting the signal to a user's specific needs (e.g. applying a frequency dependent gain, compression, feedback cancellation, etc.). Other possible devices hosting an audio enhancement device according to the invention are a headset, an active earplug, a pair of headphones, an ASR (Automatic Speech Recognition) system, etc.

[0084] FIG. 2h and 2i show embodiments of an audio enhancement device where the acoustically and wirelessly propagated input signals are processed in the frequency domain. In both embodiments, the streamed target audio signal *WIN* and the propagated electric signal *AIN* are fed to respective filter banks *FB* for splitting the input signals in a number of signals each representing a part of the frequency range of the input signal. FIG. 2h shows an embodiment where the output *OUT* from the *AS* unit representing the estimate of the target signal is synthesized to one (time dependent) signal e.g. for being presented to a user via an output transducer, whereas FIG. 2i provides the estimate of the target signal OUT from the *AS* unit in the (time) frequency domain.

[0085] FIG. 3 illustrates two embodiments of the disclosure implementing an audio enhancement device. The audio enhancement device comprises a wireless receiver in the form of an antenna and corresponding electronic circuitry adapted for picking up a wirelessly transmitted signal comprising an audio signal and demodulating it to an electrical signal representing the wirelessly received (target) audio signal transmitted (e.g. streamed) from an entertainment device, e.g. a PC or a TV-set. The acoustic signal comprising the target signal, e.g. the sound from a PC or a TV-set or a voice from a speaker or singer, and other (possibly unwanted) sources of sound or noise in the environment is picked up by a (e.g. directional) microphone system of the audio enhancement device (the embodiment of FIG. 3a

comprising a *single* microphone *m*, the embodiment of FIG. 3b comprising a *multitude* of microphones $m_1, m_2, ... m_n$). The electric input signal(s) from the microphone *m* (FIG. 3a) or the multitude of microphones $m_1, m_2, ... m_n$ (FIG. 3b) is(are) fed to corresponding filter banks *FB* (FIG. 3a) and $FB_1, FB_2, ..., FB_n$ (FIG. 3b), respectively, for converting the time variant electric input signal(s) to the (time-) frequency domain. Instead of filter banks, any other time to frequency conversion elements (e.g. Fourier Transform algorithms, such as FFT) may be used, if appropriate. The outputs from the filter bank(s) connected to the microphone(s) is(are) fed to each their adaptive filters *(FIR, LMS)* for estimating the target signal (here a FIR-filter with an LMS algorithm is indicated; other filters (e.g. IIR) and algorithms (e.g. RLS) may be used). The wirelessly received, 'clean' version of the target signal is fed to a filter bank *FB* for conversion to the (time-) frequency domain. Depending on the actual acoustic propagation distance in the room in question, the delay of the input transducer and associated circuitry and the delay of units for possible coding, error correction, etc., and on the transmission- and reception-delays of the wirelessly transmitted signal, either of the two signal paths may have the larger propagation delay from the acoustic source to the audio device (cf. e.g. FIG. 1b for an example of an acoustic and an electromagnetic propagation path). In the embodiment of FIG. 3, it is assumed that the wirelessly transmitted signal is delayed more than the acoustically transmitted signal. The amount of delay between the streamed signal and the acoustically received signal is estimated and controlled by a delay control unit *(DELAY CTRL)* receiving copies of the filter coefficients (*COEFF*) from the algorithm part(s) (*LMS*) of the adaptive filter(s) (*FIR, LMS*). The delay control unit may e.g. be implemented by an adaptive filter. The estimated target signal *OUT* is in the embodiment of FIG. 3a the output of the single variable filter part (FIR) of the single adaptive filter (FIR, LMS) and in the embodiment of FIG. 3b the output of multi input SUM unit ('+' in FIG. 3b) providing a (possibly weighted) sum of the outputs of the multitude of variable filter parts (FIR) of the multitude of adaptive filters (FIR, LMS). The estimated target signal *OUT* is e.g. fed to a signal processor (e.g. for further processing of the signal, such as applying a frequency dependent gain according to a user's needs, compression, noise reduction, etc.) and is further branched off to a delay unit (Δ), which delays the estimated target signal (*OUT*) by an estimated delay controlled by the output (*DELAY*) of the delay control unit (*DELAY CTRL*), The output of the delay unit (Δ) is subtracted (in SUM unit '+') from the wirelessly received (streamed) target signal. The delay unit can be implemented as variable delay line or as a programmable wait routine in a software algorithm. The resulting signal is fed to the algorithm part (LMS) of the adaptive filters (the multitude if algorithm parts (LMS) in the embodiment of FIG. 3b) and used to -determine (update) the filter coefficients of the adaptive filter. Thereby the wirelessly received (streamed) target signal is used to estimate the target signal extracted from the acoustically received signal and the delay problem is eliminated. The dashed (FIG. 3a), respectively solid (FIG. 3b), line enclosing units *LMS, Δ, DELAY CTRL* and a SUM-unit '+' indicates elements of the *ESTIMATOR* units of FIG. 2b-2e, 2g. The variable filter part(s) (*FIR*) of the adaptive filter(s) (*LMS, FIR*) (and the sum unit '+' in FIG. 3b connected to the outputs of the variable filter parts) represent(s) the *ACTUATOR* unit of FIG. 2b-2e, 2g.

[0086] The proposed scheme can be used to correct or re-establish the audio characteristics ('audio fingerprints') of the received acoustic signal in accordance with the target signal, e.g. spectral, temporal and modulation characteristics (such as pitch, onset, offset, cepstral coefficients, MFCC (Mel Frequency Cepstral Coefficients, etc.). Further, the proposed scheme can be used to compensate for non-linearities in loudspeakers in a room (so that the resulting version of the sound signal is NOT 'destroyed' by bad components). Such characteristics can e.g. be extracted from the wirelessly received target signal (possibly in combination with corresponding characteristics extracted from the propagated electric signal) in the *ESTIMATOR* unit and applied to the estimate of the target signal in the *ACTUATOR* unit by means of control signals *CTR* (cf. e.g. FIG. 2b, 2c, 2g, 3).

[0087] FIG. 4 shows a listening device comprising an audio enhancement device according to an embodiment of the invention. The listening device 400 (*LD*), e.g. a hearing instrument, comprises an audio enhancement device (*AE*) 40 (enclosed by the dotted rectangle), a signal processing unit *(DSP)* 48, a digital to analogue converter (DA) 49 and an output transducer 50 (here a receiver). The audio enhancement device 40 comprises a microphone system 41 (depicted as a single microphone but in practice possibly comprising a multitude of microphones) for converting an input sound comprising a target sound and a noise signal (cf. *Acoustic Target+Noise* in FIG. 4) to an *(propagated)* electric input signal *AIN* comprising said target signal and said noise signal, an analogue to digital converter *(AD)* 42 for providing a digitized electric input signal comprising said target signal and said noise signal. The digitized electric input signal is fed to the audio enhancement unit (AS) 47, as e.g. depicted in more detail in FIG. 2b-2e, 2g and 3a-3b (comprising the *ESTIMATOR* and *ACTUATOR* blocks in FIG. 2 and their equivalents in FIG. 3). The audio enhancement device 40 further comprises a wireless receiver (comprising antenna 44 and receiver unit (Rx)- 45) for receiving a signal (cf. *ElectroMagnetic Target* in FIG. 4) comprising the target audio signal and for extracting said (*streamed*) target audio signal. The target audio signal is digitized in analogue to digital converter (*AD*) 46. The digitized output of the AD-converter comprising the wirelessly received target audio signal is fed to the audio enhancement unit *(AS)* 47. The signal processing unit 48 *(DSP)* is adapted to process the output signal *OUT* from the audio enhancement unit *(AS),* e.g. to adapt the signal to a specific user's hearing profile (including applying a frequency dependent gain). The signal from the signal processing unit 48 is connected to the DA-converter 49, whose analogue output is fed to the receiver 50 for presenting an *Enhanced output* to a user. The hearing instrument may further comprise other circuitry for improving the signal presented to the user, e.g. an anti-feedback system.

[0088] FIG. 5 shows an example of an audio enhancement system according to an embodiment of the disclosure, the system comprising an audio enhancement device wherein the target signal is estimated based on the acoustically propagated signal based on LMS deconvolution. The embodiment of an audio enhancement system shown in FIG. 5 entertainment device 30 comprising an audio source (comprising electrical target signal S and speaker 31 for converting the target

signal S to an acoustic target signal), and a transmitting device (comprising transmitter (*Tx*) 33 and antenna 32) for generating a wireless signal comprising a representation of the target signal in the form of a target audio signal. The audio enhancement system further comprises a receiving device 40 comprising an audio enhancement device (*AE*). The acoustic target signal generated by the speaker 31 follows an acoustic propagation path (cf. arrow denoted *AC* D_A, *H* in FIG. 5) from the speaker 31 to the microphone 41 of the audio enhancement device 40. Along the acoustic propagation path, the acoustically propagated target signal is modified, including being delayed by an amount D_A and subject to a (typically frequency dependent) transfer function *H.* Also other acoustic source signals in the environment are added along the acoustic propagation path (cf. arrow denoted *AC N)* in FIG. 5. Likewise, the wireless signal generated by the transmitter 33, 32 follows an electromagnetic propagation path (cf. arrow denoted *EM* D_{EM} in FIG. 5) from the transmitter (33, 32) of the entertainment device 30 to the receiver (44, 45) of the audio enhancement device 40. The audio enhancement device comprises a microphone system 41 for converting the acoustically propagated signal from the speaker 31 to an electric input signal and an

[0089] AD-converter 42 for sampling_the electric input signal and providing a *digitized* input signal 421, which is fed to an adaptive system 48 (H_{est}), e.g. an adaptive filter, for providing as an output *OUT* an estimate of the target signal. The audio enhancement device 40 further comprises a wireless receiver (comprising antenna 44 and receiver unit (Rx) 45) for receiving a signal comprising the target audio signal and for extracting said (*streamed*) target audio signal. The target audio signal is digitized in analogue to digital converter (*AD*) 46, whose output the digitized streamed target audio signal 461 is fed to a delay estimate unit 47 (ΔD_{est}) together with the digitized propagated electric signal 421, the delay estimate unit 47 being adapted for estimating the difference (D_A - D_{EM}) in delay between the two input signals and providing as an output 471 a digitized streamed target audio signal delayed by D_A-D_{EM} (here D_A is assumed to be larger than D_{EM}; if this is not the case, the order of the delays should be reversed). The delayed digitized streamed target audio signal 471 is subtracted from the estimate of the target signal *OUT* in SUM unit 49 (+). The resulting signal is used in the adaptive system 48 (H_{est}) for estimating the acoustic propagation path H resulting in a transfer function providing an estimate of H^{-1} (H_{est}^{-1}). The wirelessly propagated signal received at the receiver is delayed by D_{EM} (D_{EM} possibly including delays in Tx, Rx, and AD units) and can be written as $S \cdot Z^{-DEM}$. The acoustically propagated signal picked up by the microphone 41 can be written as $S \cdot z^{-DA} \cdot H + N$ (D_A possibly including delays in microphone and AD-conversion units). The delay estimate unit 47 (ΔD_{est}) estimates the delay difference (D_A - D_{EM}) providing the transfer function $Z^{-(DA-DEM)}$ resulting in an output signal 471 of the ΔD_{est} unit of the form $S \cdot z^{-DA}$. The delay difference (D_A - D_{EM}) may e.g. be determined-by considering the cross correlation between the two input signals. The resulting output OUT of the adaptive system 48 can thus be written as

$$S \cdot z^{-D_A} \cdot H \cdot H_{est}^{-1} + N \cdot H_{est}^{-1}.$$

If H_{est} is a good estimate of H, the output OUT

$$(OUT = S \cdot z^{-D_A} + N \cdot H_{est}^{-1})$$

of the signal enhancement unit comprises the estimate of the target signal plus a

noise contribution coloured by the estimate H_{est} of the acoustic propagation path (comprising contributions from the environment, e.g. reflections of a room and its contents, the loudspeaker 31, the microphone 41, etc.). The smaller the noise contribution N, the better the target signal estimate. If a noise component N_i contributing to the total noise N is constant (or periodic), e.g. originating from a fan, a dish washing machine or the like, such contribution may be filtered out, thereby improving the target signal estimate.

[0090] FIG. 6 shows an audio enhancement system according to an embodiment of the invention, the system comprising an audio enhancement device wherein the target signal is estimated based on the electromagnetically propagated signal. The embodiment of an audio enhancement system of FIG. 6 resembles the embodiment of FIG. 5. The roles of the acoustically and wirelessly propagated signals in the estimation of the target signal are, however, switched. The entertainment device 30 (e.g. an A/V-device, such as a TV) and the propagation scenario is assumed to be equivalent to that of FIG. 5. In the following, only the receiving device 60 comprising an audio enhancement device (*AE*) is described. The audio enhancement device (*AE*) comprises a microphone system 61 for converting the acoustically propagated signal from the speaker 31 to an electric input signal and an AD-converter 62 for sampling the electric input signal and providing a *digitized* input signal 621, which is fed to a delay estimate unit 67 (ΔD_{est}) as well as to SUM unit 69 ('+). The audio enhancement device 60 further comprises a wireless receiver (comprising antenna 64 and receiver unit (*Rx*) 65) for receiving a signal comprising the target audio signal and for extracting said (*streamed*) target audio signal. The target audio signal is digitized in analogue to digital converter (*AD*) 66, whose output the digitized streamed target audio signal 661 is fed to the delay estimate unit 67 (ΔD_{est}) together with the digitized propagated electric signal 621, the delay estimate unit 67 being adapted for estimating the difference (D_A-D_{EM}) in delay between the two input signals and providing as an output 671 a digitized streamed target audio signal delayed by D_A-D_{EM} (here D_A is assumed to be larger than D_{EM}; if this is not the case, the order of the delays should be reversed). The delayed digitized streamed target audio signal 671 is fed to an adaptive system 68 (H_{est}) for providing as an output *OUT* an estimate of the target signal. The output OUT of the adaptive system (e.g. an adaptive filter, e.g. a FIR filter) is subtracted from the digitized propagated electric signal 621 in SUM unit 69 (+). The resulting signal is used in the adaptive system 68 (H_{est}), e.g. implemented using an adaptive filter, for estimating the acoustic propagation path H resulting in a transfer function providing an estimate of H (H_{est}). The wirelessly propagated signal received at the receiver is delayed by D_{EM} (D_{EM} possibly including delays in Tx, Rx, and AD units) and can be written as $S \cdot z^{-DEM}$. The acoustically propagated signal picked up by the microphone 61 can be written as $S \cdot z^{-DA} \cdot H + N$ (D_A possibly including delays in microphone and AD-conversion units). The delay estimate unit 67 (ΔD_{est}) estimates the delay difference (D_A - D_{EM}) providing the transfer function $z^{-(DA-Dem)}$ resulting in an output signal 671 of the ΔD_{est} unit of the form $S \cdot z^{-DA}$. The delay difference (D_A - D_{EM}) may e.g. be determined by considering the cross correlation between the two input signals. The output of the SUM unit 69 used for estimating the acoustic propagation path H in the adaptive system 68 can be written as $S \cdot z^{-DA} \cdot H + N - S \cdot$

$z^{-DA} \cdot H_{est.}$ The resulting output OUT of the adaptive system 68 can be written as $OUT = S \cdot z^{-DA} \cdot H_{est.}$ If H_{est} is a good estimate of H, the output *OUT* of the signal enhancement unit comprises the estimate of the target signal coloured by the estimate H_{est} of the acoustic propagation path (comprising contributions from the environment, e.g. reflections of a room and its contents, head-related transfer functions HRTF, the loudspeaker 31, the microphone 41, etc.). The colouring of the otherwise 'clean', wirelessly transmitted target signal can e.g. be of interest in specific situations. In a preferred embodiment, the above described way of estimating the target signal based on the wirelessly propagated signal is used in specific listening situations, where an impression of the room or other acoustic environment (e.g. a concert situation) is of importance, is e.g. implemented in a specific program of a listening device, e.g. a hearing instrument, such program being e.g. adapted to be switched on and off by the user. In an embodiment comprising two listening devices, one located at or in each ear of a user, the above scheme provides the possibility to implement individualized HRTFs. In an embodiment, the adaptive system 68 (H_{est}) for estimating the acoustic propagation path H comprises an adaptive filter of an order higher than 60, such as higher than 120, such as higher than 240.

[0091] FIG. 7 shows block diagrams of two embodiments of a listening device according to the disclosure, the listening devices comprising an audio enhancement device (AE) electrically connected to a signal processing

[0092] unit (*DSP*) and a speaker/receiver, said devices comprising a forward path for a target audio signal. The embodiments of the audio enhancement device (*AE*) of FIG. 7a and 7b comprise the same elements as the embodiments shown in FIG. 2b and 2c, respectively. Additionally, the embodiments of an audio enhancement device (*AE*) of FIG. 7a and 7b comprise (as in FIG. 2f) an output signal *CTR2* comprising characteristics of the target signal extracted from the *streamed* target audio input signal and/or from the *propagated* electric signal for use as an input to the signal processing unit (*DSP*). FIG. 7a shows an embodiment where the target signal is estimated based on the acoustically propagated signal (*AIN*) and characteristics of the target signal are extracted from the electromagnetically propagated signal (*WIN*) and used (signal *CTR1*) in the estimate of the target signal *OUT.* FIG. 7b shows an embodiment where the target signal is estimated based on the electromagnetically propagated signal (*WIN*) and characteristics of the target signal are extracted from the acoustically propagated signal (*AIN*) and used (signal *CTR1*) in the estimate of the target signal *OUT.* The listening devices of FIG. 7 may form part of a communication device, such as a mobile telephone or a listening device, such as a hearing instrument. In an embodiment, digital processing parts of the audio enhancement unit (AE), e.g. the estimator (*E*) and actuator (*A*) units, form part of a digital signal processor. In an embodiment, some or all of the functions of the estimator (*E*) and actuator (*A*) units are implemented as software algorithms.

[0093] FIG. 8 shows an audio enhancement system adapted for enhancement of a particular voice according to an embodiment of the disclosure. The embodiment of an audio enhancement system shown in FIG. 8 comprises at least two listening

devices, e.g. hearing instruments (or pairs of listening devices, e.g. hearing instruments), a first device *HA1* being worn by a first user whose voice is acting as an audio source for generating an acoustic target signal *ow1* and, the first device *HA1* comprising a transmitting device (*Tx* and antenna in *HA1*) for generating a wireless signal *ow1-em(D_{EM})* comprising a representation of said target signal *ow1* in the form of a streamed target audio signal. The audio enhancement system further comprises a second listening device *HA2* (cf.e.g. described in connection with FIG. 7a) worn by a second user and comprising an audio enhancement device comprising a receiving device (antenna in *HA2* and other receive and demodulation circuitry, not shown). The acoustic propagation path from the acoustic source of the target signal *ow1* (user one) to the listening device *HA2* worn by user 2 is characterized by transfer function *H* and delay D_A resulting in signal *ow1(H, D_A)* received by *HA2* (here neglecting possible noise N added to the acoustic signal). The transmitting device *HA1* comprises a microphone for picking up an input sound, here a user's voice *ow1* and converting the input sound to an analogue electric input signal, which is digitized in an analogue to digital converter *AD* whose digitized output is fed to a processing unit *DSP* comprising an own voice detector *OWD* for detecting and extracting a user's own voice (this can e.g. be implemented as described in WO 2004/077090 A1 or in EP 1 956 589 A1). A signal comprising the user's own voice (or characteristics thereof) is fed to a transmitter *Tx* and wirelessly transmitted to the receiving device *HA2* (via antennas of the transmitting and receiving devices). The wireless propagation path of the target signal *ow1-em* from the transmitting device *HA1* to the receiving device *HA2* is characterized by a delay D_{EM}, resulting in signal *ow1-em(D_{EM})* being received by *HA2.* The audio enhancement unit *AE* of *HA2* is adapted to estimate the target signal *ow1* (the voice of user 1) based on the received wirelessly streamed signal *ow1-em(D_{EM})* (as e.g. described in connection with FIG. 5), whereby the output signal *OUT* of the audio enhancement unit of *HA2* comprises an enhanced version of the acoustically propagated voice of user 1 (target signal). The estimated target signal *(OUT)* is fed to a signal processing unit *DSP* for possible further processing of the signal and eventual presentation of an enhanced (user adapted) output signal to user 2 via an output transducer (here receiver). A further signal CTR from the audio enhancement unit comprising characteristics of the target signal is fed to the signal processing unit and used in the further processing of the estimated target signal *(OUT).* The forward path of the listening device *HA1* worn by user 1 comprises a number of electrically connected elements including - in addition to the microphone, *AD*-converter and signal processing unit (*DSP*) - a digital to analogue converter (*DA*) and a receiver for converting an electric signal to an output sound for being presented to user 1. In an embodiment, the first listening device *HA1* comprises an audio enhancement unit *AE* as illustrated in *HA2.* In an embodiment, both listening devices *HA1* and *HA2* are essentially identical, e.g. so that *HA1* is specifically adapted to estimate the voice of user 2, e.g. as described above for *HA2.*

[0094] FIG. 9 shows a flow diagram of a method of enhancing an audio signal in a receiving device according to an embodiment of the present invention. The method comprises

S1. Acoustically propagating a target signal from an acoustic source along an acoustic propagation path, providing a propagated acoustic signal at the receiving device;

S2. Converting the received propagated acoustic signal to a *propagated* electric signal, the received propagated acoustic signal comprising the target signal, noise and possible other sounds from the environment as modified by the propagation path from the acoustic source to the receiving device;

S3. Wirelessly transmitting a signal comprising the target audio signal to the receiving device;

S4. Receiving the wirelessly transmitted signal in the receiving device;

S5. Retrieving a *streamed* target audio signal from the wirelessly received signal comprising the target audio signal;

S6. Estimating the target signal from the *propagated* electric signal and the *streamed* target audio signal using an adaptive system or algorithm.

[0095] Preferably, at least some of the steps of the method are implemented as software algorithms. In an embodiment, at least step 6 (S6) is implemented as one or more software algorithms. Preferably such software algorithms are adapted for running on a signal processing unit of the receiving device, e.g. a listening device, e.g. a hearing instrument.

[0096] FIG. 10 shows an example of characteristics of the streamed target audio signal, here modulation index MI vs. time t (FIG. 10a) and resulting inputs to a processing algorithm providing gain G [dB] vs. modulation index MI. FIG. 10a schematically shows a voice signal input (amplitude A vs. time t) for a wirelessly received target audio signal (*Streamed audio signal*). The signal varies between a bottom tracker level (*BT*) and a top tracker level (*TT*). The modulation index (MI) indicates the ratios of (difference between in a dB-terms) the top tracker level and the bottom tracker level. The bottom tracker can be taken as an estimate of the noise floor, $N_{est.}$, whereas the top tracker can be taken as an estimate of the signal (S) plus noise (N), $(S+N)_{est.}$ Various aspects of noise reduction using the modulation index or modulation amplitude are discussed in WO 2005/086536 A1. These characteristics of the streamed target audio signal can be used in a voice controlled noise reduction algorithm to scale gain of the propagated electric signal as indicated in FIG. 10b, where gain (G [dB]) vs. modulation index (MI) is schematically shown. For values of the modulation index MI below a value MI1 (e.g. 0.5), gain G applied to the propagated electric signal is constant G1 [dB] (e.g 12 dB), whereas applied gain decreases linearly (in dB) for values of MI larger than MI1. The appropriate values of G1 and MI1 may vary depending on the acoustic environment, the specific listening device, etc. The characteristics of FIG. 10 can e.g. be used in an embodiment of an audio enhancement device as shown in FIG. 2d, where the estimator unit (E) comprises a voice detector for detecting a voice in the propagated electric signal and an algorithm for extracting a modulation

index and determining a corresponding gain, and an activator unit (A) comprising an algorithm for correspondingly applying the appropriate gain to the propagated electric signal.

[0097] FIG. 11 shows an example of a comparison of characteristics of the streamed target audio signal with those of the acoustically propagated signal. FIG. 11 a schematically shows a voice signal input (amplitude A vs. the t) for a wirelessly received streamed target audio-signal (*Streamed audio signal,* dashed graph) and for the corresponding propagated electric signal (*Acoustically propagated signal,* solid graph). The signals vary between a bottom tracker level (*BT*) and a top tracker level (*TT*), here denoted BT-S, TT-S and BT-A, TT-A for the streamed target audio signal and the propagated electric signal, respectively. The modulation indices MI-S and MI-A for the streamed target audio signal and the propagated electric signal, respectively, are indicated. FIG. 11b and 11c show resulting inputs to processing algorithms based on the respective top and bottom tracker data. FIG. 11b shows data for incremental gain ΔG [dB] vs. frequency f extracted as the difference between the corresponding top tracker (TT) vs. frequency data for the *Streamed audio signal* (dashed graph) and the *Acoustically propagated signal* (solid graph). These data are used as input to an algorithm for estimating signal gain. FIG. 11c shows data for incremental noise reduction ΔNR [dB] vs. frequency f extracted as the difference between the corresponding bottom tracker (BT) vs. frequency data for the *Streamed audio signal* (dashed graph) and the *Acoustically propagated signal* (solid graph). These data are used as input to an algorithm for estimating noise reduction. The characteristics of FIG. 11 can e.g. be used in an embodiment of an audio enhancement device as shown in FIG. 2h or 2i, where the audio enhancement unit (*AS*) comprises an estimator unit (E) comprising an algorithm for detecting top and bottom trackers vs. frequency of the streamed target audio signal as well as for the propagated electric signal and for providing data for the frequency dependence of the difference between the top trackers and between the bottom trackers for the streamed target audio signal and the propagated electric signal, respectively, and an activator unit (A) comprising algorithm(s) for correspondingly applying the appropriate gains to the appropriate parts of the propagated electric signal.

[0098] FIG. 12 shows an audio enhancement device according to an embodiment of the disclosure (FIG. 12A) and corresponding characteristics of the target signal, here level differences ΔG [dB] vs. frequency f (FIG. 12b). FIG. 12a shows an audio enhancement device (*AE*) comprising an antenna and corresponding receiver and demodulation circuitry for receiving (and demodulating) a wirelessly transmitted signal comprising a target audio signal, the receiver being adapted for providing as an output a streamed target signal WIN. The streamed target signal WIN is connected to a time to frequency conversion unit, here a filter bank *FB*. The filter bank splits the input signal *WIN* into a number P of time variant signals (WINp=WIN1, *WIN2, ..., WINP*), each comprising a separate frequency range or band of the input signal. Likewise, the audio enhancement device comprises a microphone for picking up the acoustically propagated signal comprising a target signal and providing a propagated electric signal *AIN,* which is connected to a time to frequency conversion unit, here a filter bank *FB.* The filter bank splits the input

signal AIN into a number P of time variant signals (AINp=*AIN1, AIN2, ..., AINP*), each comprising a separate frequency range or band of the input signal. The time-frequency domain signals WINp and AINp are connected to each their level detection units (*LD*) for detecting an input level of the individual signal components WINp and AINp. The outputs LD-wp (=*LD-w1, LD-w2, ..., LD-wP*) and LD-ap (=*LD-a1, LD-a2, ..., LD-aP*) representing, respectively, input level vs. frequency f (or index p, p=1, 2, ..., P) of the streamed target audio signal and the propagated electric signal are connected to a processing unit (C) for comparing the respective input levels at different frequencies, cf. FIG. 12b depicting LD-w(f) (or corresponding values of LD-wp and p) (dashed graph) and LD-a(f) (or corresponding values of LD-ap and p) (solid graph). Various aspects of level detection in a listening device is e.g. discussed in WO 2003/081947 A1. The two level detection units (LD) and the processing unit C together form part of an estimation unit (E), as indicated by the dashed outline enclosing these units. The processing unit calculates e.g. values $\Delta G(f)=LD\text{-}w(f) - LD\text{-}a(f)$ at different frequencies (e.g. $\Delta G(p)=LD\text{-}wp - LD\text{-}ap$, p= 1, 2, ..., P). These values are fed to actuator unit (A) as signals Cp=(*C1, C2, ..., CP*), where they are used as inputs to an algorithm for modifying the gain of the propagated electric signal *AINp,* which are inputs to the actuator unit (taken from the corresponding filter bank *FB*). The output *OUT* of the actuator *unit A* provides an enhanced estimate of the target signal. The output *OUT* may be either in a time domain for being further processed or presented directly to a user via an output transducer or in the time frequency domain adapted for being subject to further processing in this framework.

[0099] The invention is defined by the features of the independent claim(s). Preferred embodiments are defined in the dependent claims. Any reference numerals in the claims are intended to be non-limiting for their scope.

[0100] Some preferred embodiments have been shown in the foregoing, but it should be stressed that the invention is not limited to these, but may be embodied in other ways within the subject-matter defined in the following claims.

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Patentkrav

1. Fremgangsmåde til forbedring af et audiosignal i en modtageanordning (60), der omfatter

• (S1) akustisk udbredelse af et målsignal fra en akustisk kilde (S, 31) langs en akustisk udbredelsesvej, der tilvejebringer et udbredt akustisk signal ved modtageanordningen (60);

• (S2) konvertering af det modtagne udbredte akustiske signal til et udbredt elektrisk signal *(AIN;* 621), hvor det modtagne udbredte akustiske signal omfatter målsignalet, støj og eventuelle andre lyde fra omgivelserne, som er modificeret af udbredelsesvejen fra den akustiske kilde til modtageanordningen (60);

• (S3) trådløs transmission af et signal, der omfatter en ren version af målaudiosignalet til modtageanordningen (60);

• (S4) modtagelse af det trådløse transmitterede signal i modtageanordningen (60);

• (S5) genvinding af et streamet målaudiosignal *(WIN;* 661) fra det trådløst modtagne signal, der omfatter den rene version af målaudiosignalet;

• (S6) estimering af målsignalet fra det udbredte elektriske signal *(AIN;* 621) og det streamede målaudiosignal *(WIN;* 661) ved anvendelse af et adaptivt system (A; 68), hvor det adaptive system omfatter en adaptiv algoritme; KENDETEGNET VED, AT fremgangsmåden yderligere omfatter

• estimering af målsignalet *fra det streamede målaudiosignal (WIN;* 661) ved anvendelse af det udbredte elektriske signal *(AIN;* 621) eller et signal udledt derfra som et input til den adaptive algoritme for at forbedre estimatet af målsignalet, og

• uddragning af egenskaber af den akustiske udbredelsesvej fra det udbredte akustiske signal, hvor uddragede egenskaber af den

akustiske udbredelsesvej anvendes til at tilføje rumlig information til målsignalestimatet (*OUT*).

2. Fremgangsmåde ifølge krav 1, der omfatter estimering af forsinkelsesforskellen mellem det udbredte elektriske signal og det streamede målaudiosignal eller mellem signaler, der stammer derfra.

3. Fremgangsmåde ifølge krav 2, der omfatter anvendelse af den resulterende forsinkelsesforskel i estimeringen af målsignalet.

4. Fremgangsmåde ifølge et hvilket som helst af kravene 1-3, der omfatter udførelse af i det mindste noget af signalbehandlingen, der er forbundet med forbedring af audiosignalet i modtageanordningen i separate frekvensområder eller -bånd.

5. Fremgangsmåde ifølge et hvilket som helst af kravene 1-4, der omfatter uddragning af egenskaber af målsignalet fra det streamede målaudiosignal.

6. Fremgangsmåde ifølge krav 5, hvor egenskaberne af målsignalet indbefatter en eller flere af de følgende: frekvensspektrummet, modulation ved forskellige frekvenser, for eksempel top/bund følgere af et modulationsindeks, onset/offset egenskaber, inputniveau.

7. Fremgangsmåde ifølge krav 5 eller 6, hvor de uddragede egenskaber af det streamede målaudiosignal anvendes som inputs til behandling af algoritmer, for eksempel forstærknings- eller støjreduktionsalgoritmer, til forbedring af målsignalet.

8. Fremgangsmåde ifølge et hvilket som helst af kravene 1-7, hvor egenskaberne af den akustiske udbredelsesvej indbefatter en eller flere af de følgende: retningsinformation, interaurale forskelssignaler, afstandsinformation, intensitet, direkte til efterklangsenergiforhold, rumindtryk.

9. Fremgangsmåde ifølge et hvilket som helst af kravene 1-8, hvor det udbredte akustiske signal er dæmpet, for eksempel annulleret, i eller ved modtageanordningen, før det præsenteres til en bruger.

10. Fremgangsmåde ifølge et hvilket som helst af kravene 1-9, der anvendes i en beskyttelsesanordning, en hovedtelefon eller et headset, et høreapparat eller et par af høreapparater til en binaural udrustning.

11. Audioforbedringsanordning (*AE*) til at forbedre et audiosignal, der omfatter

i det mindste en indgangstransducer (*m*; 61) til konvertering af et udbredt akustisk signal, der omfatter et målsignal, som er udbredt fra en akustisk kilde langs en akustisk udbredelsesvej til audioforbedringsanordningen til et *udbredt* elektrisk indgangssignal *(AIN;* 621);

en trådløs modtager (64, 65) til modtagelse af en ren version af et målaudiosignal via en trådløs forbindelse, og som tilvejebringer et *streamet* målaudiosignal *(WIN;* 661);

et første adaptivt system (A; 68) til estimering af målsignalet, der er baseret på det u*dbredte* elektriske indgangssignal *(AIN;* 621) og det *streamede* målaudiosignal *(WIN;* 661), hvor det første adaptive system omfatter en adaptiv algoritme, KENDETEGNET VED, AT

det adaptive system (68) er konfigureret til at estimere målsignalet *fra det streamede målaudiosignal* ved anvendelse af det udbredte elektriske signal eller et signal, der er udledt derfra som et input til den adaptive algoritme, OG VED AT

audioforbedringsanordningen yderligere omfatter en estimatorenhed (*E*) til uddragning af egenskaber af den akustiske udbredelsesvej fra det udbredte akustiske signal, hvor uddragede egenskaber af den akustiske udbredelsesvej anvendes til at tilføje rumlig information til målsignalestimatet (*OUT*).

12. Audioforbedringsanordning ifølge krav 11, hvor det første adaptive system omfatter et adaptivt filter (68) til tilvejebringelse af estimatet (*OUT*) af målsignalet, hvor det adaptive filter (68) omfatter en variabel filterdel og en opdateringsalgoritmedel, hvor opdateringsalgoritmedelen omfatter den adaptive algoritme, hvor den variable filterdel tilvejebringer en overføringsfunktion (H_{est}), der er indstillet til automatisk at ændre input, der er baseret på den adaptive algoritme af opdateringsalgoritmedelen.

13. Audioforbedringsanordning ifølge krav 11 eller 12, der omfatter en udgangstransducer til at præsentere estimatet af målsignalet eller et output fra en signalbehandlingsenhed, der omfatter en yderligere behandling af estimatet af målsignalet som et stimulus, der er indrettet til at blive opfattet af en bruger som en udgangslyd.

14. Audioforbedringsanordning ifølge krav 13, der omfatter en udgangstransducer til et øresneglsimplantat eller til en knogleledende-høreanordning.

15. Audioforbedringssystem, der omfatter en audiokilde (S, 31) til frembringelse af et akustisk målsignal og en transmissionsanordning (32, 33) til frembringelse af et trådløst signal, der omfatter en ren fremstilling af målsignalet i form af et målaudiosignal og en modtageanordning (60), der omfatter en audioforbedringsanordning (*AE*) ifølge et hvilket som helst af kravene 10-14.

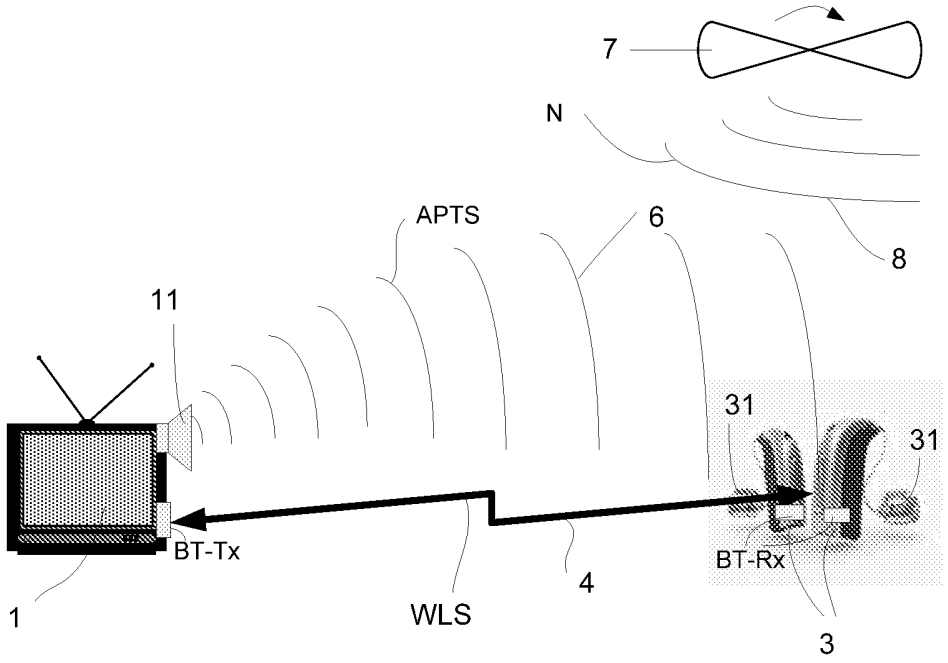

7
N
APTS
6
8
11
31
31
BT-Tx
BT-Rx
4
1
WLS
3

FIG. 1a

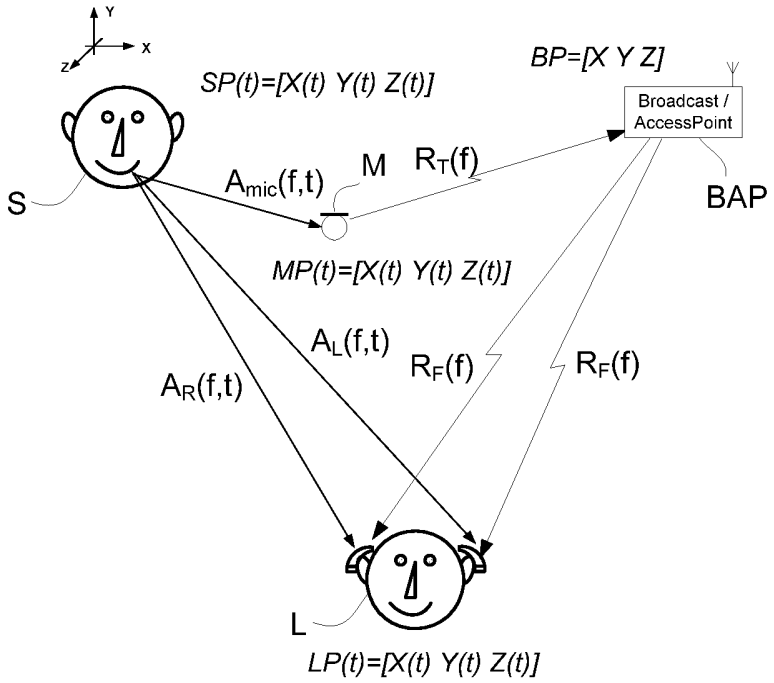

Y
X
Z
SP(t)=[X(t) Y(t) Z(t)]
BP=[X Y Z]
Broadcast / AccessPoint
S
M
$R_T(f)$
$A_{mic}(f,t)$
BAP
MP(t)=[X(t) Y(t) Z(t)]
$A_L(f,t)$
$R_F(f)$
$R_F(f)$
$A_R(f,t)$
L
LP(t)=[X(t) Y(t) Z(t)]

FIG. 1b

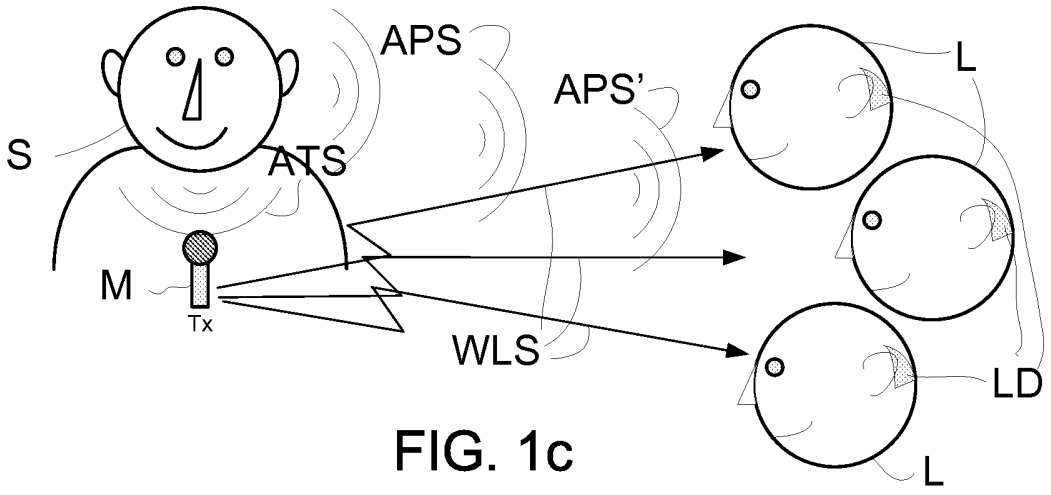
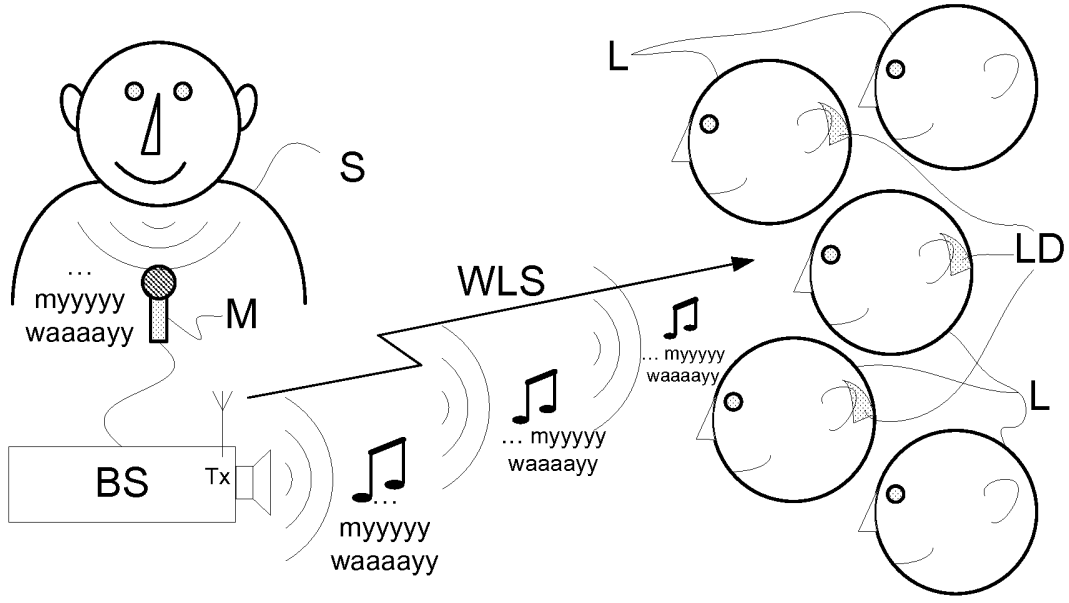

APS
APS'
S
ATS
L
M
Tx
WLS
LD
L

FIG. 1c


L
S
LD
...
myyyyy
waaaayy
M
WLS
... myyyyy
waaaayy
... myyyyy
waaaayy
L
BS
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waaaayy

FIG. 1d

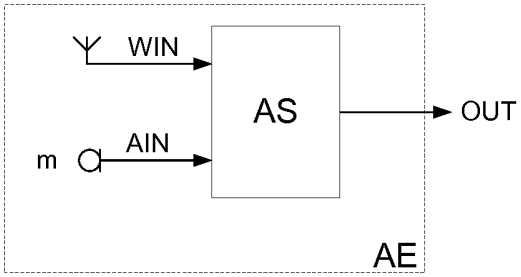

FIG. 2a

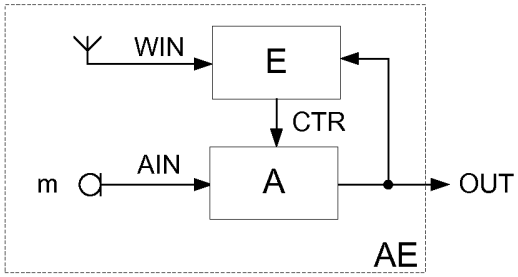

FIG. 2b

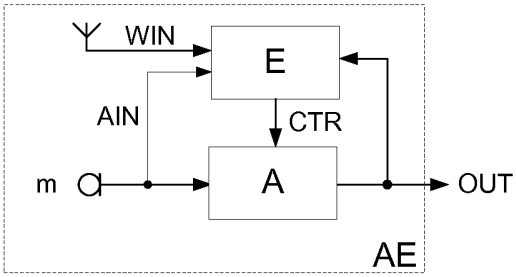

FIG. 2c

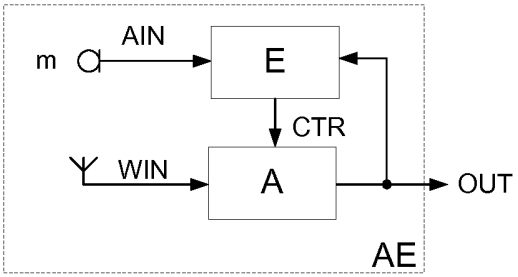

FIG. 2d

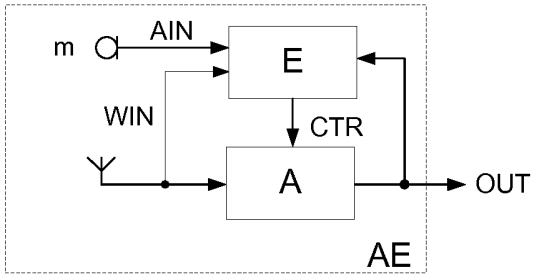

FIG. 2e

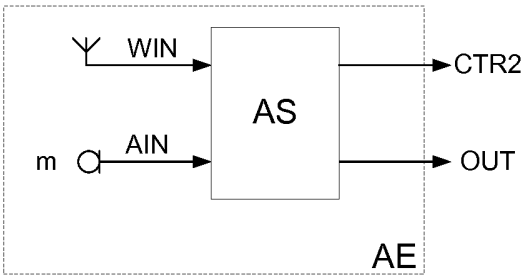

FIG. 2f

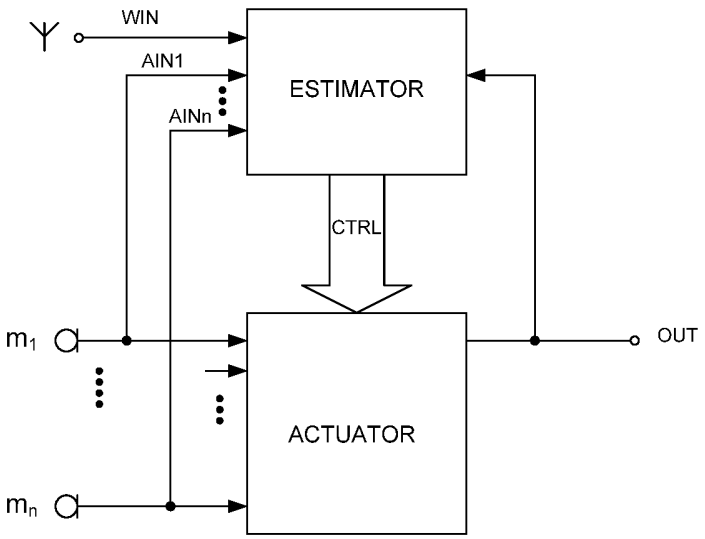

FIG. 2g

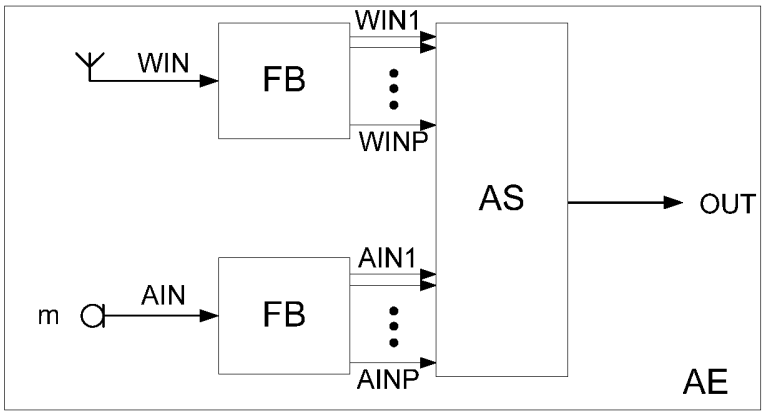

FIG. 2h

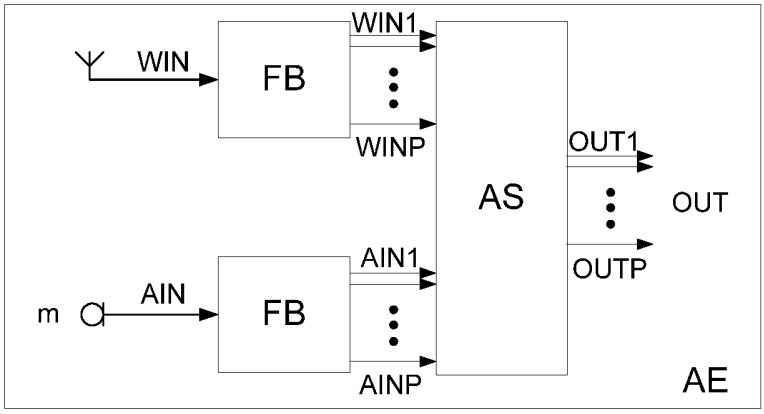

FIG. 2i

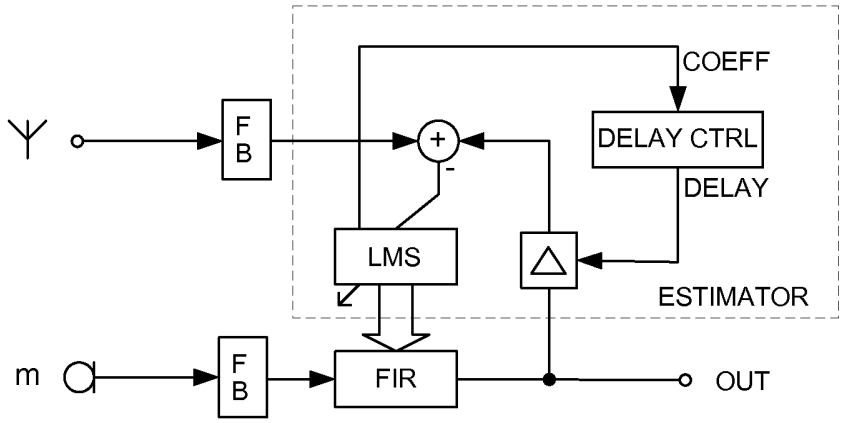
COEFF
DELAY CTRL
DELAY
F
B
LMS
ESTIMATOR
m
F
B
FIR
OUT

FIG. 3a

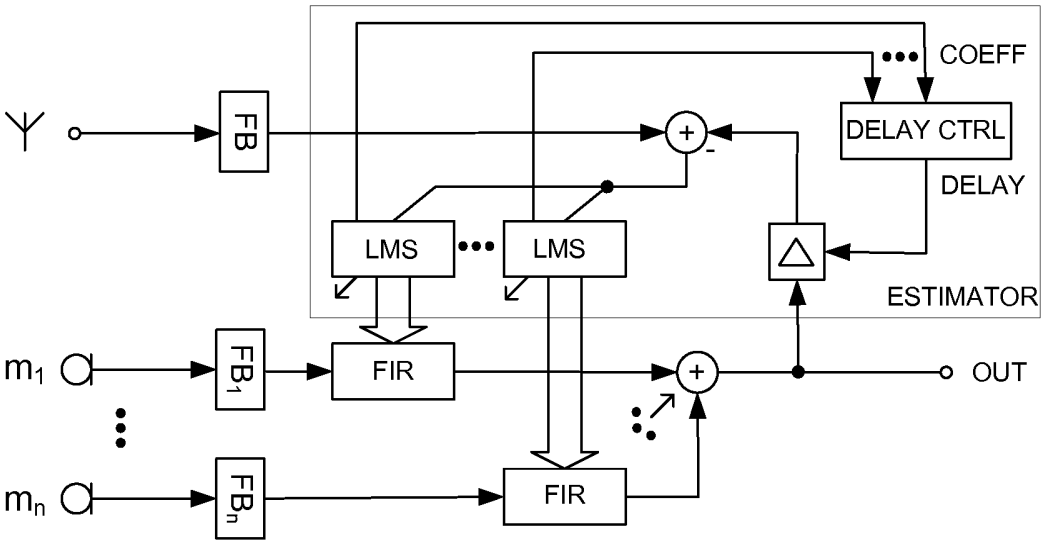
COEFF
DELAY CTRL
DELAY
FB
LMS
LMS
ESTIMATOR
m1
FB1
FIR
OUT
mn
FBn
FIR

FIG. 3b

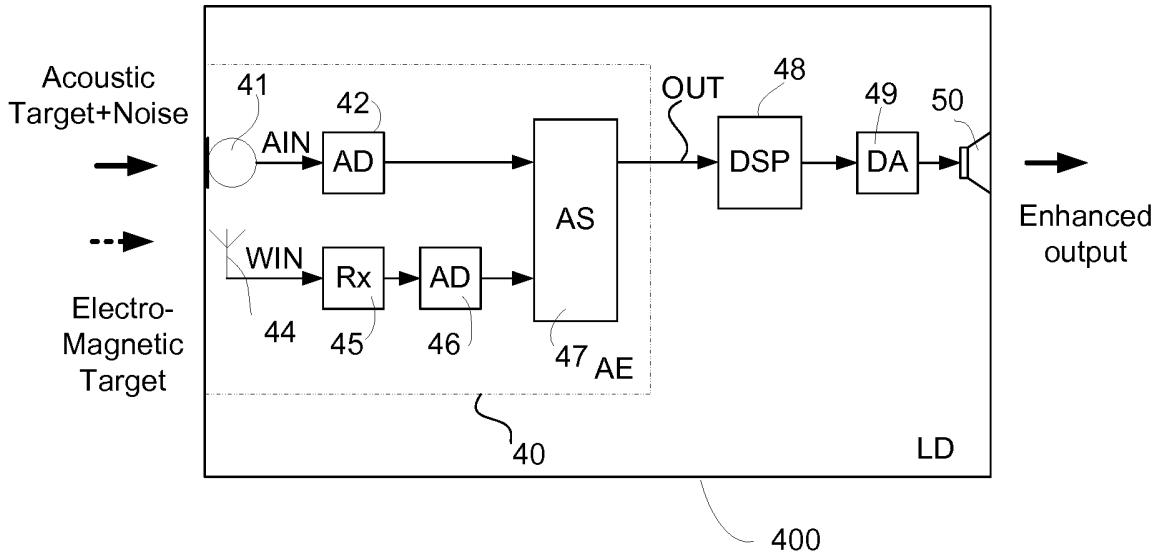
Acoustic
Target+Noise
41
42
AIN
AD
OUT
48
49
50
DSP
DA
Enhanced
output
AS
WIN
Rx
AD
Electro-
Magnetic
Target
44
45
46
47
AE
40
LD
400

FIG. 4

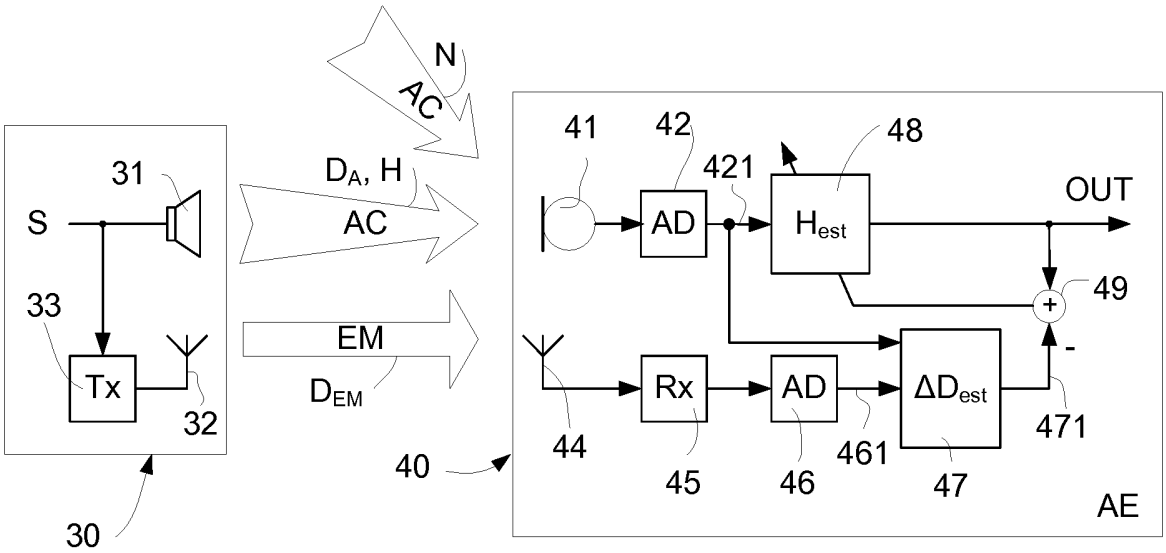
N
AC
31
S
33
Tx
32
30
D_A, H
AC
EM
D_{EM}
40
41
42
421
48
AD
H_{est}
OUT
49
+
-
Rx
AD
ΔD_{est}
471
44
45
46
461
47
AE

FIG. 5

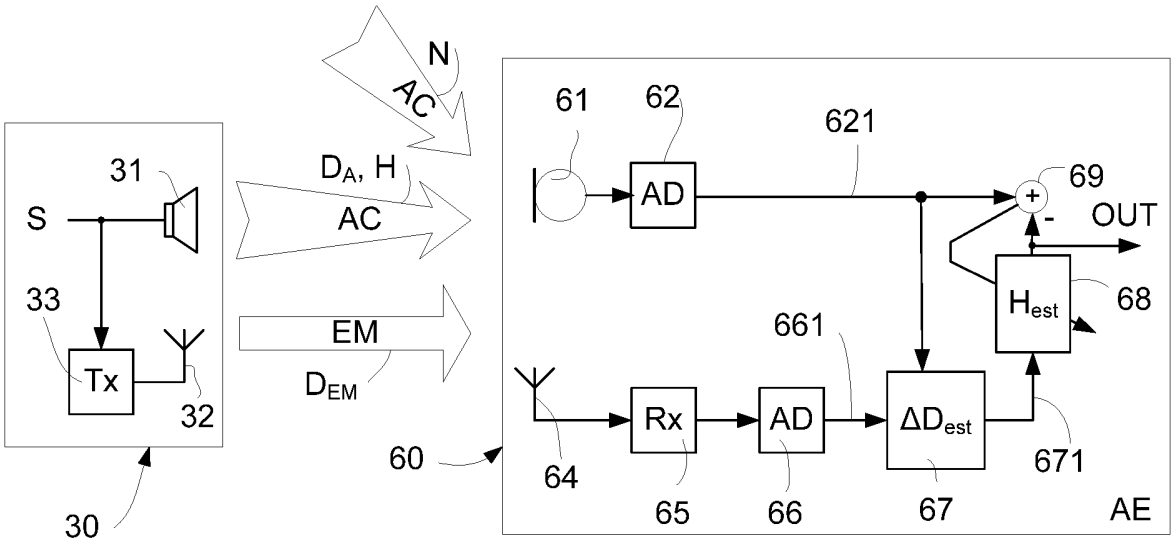

FIG. 6

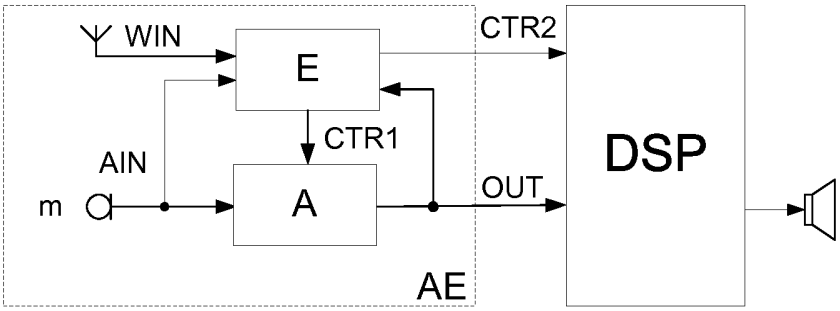

FIG. 7a

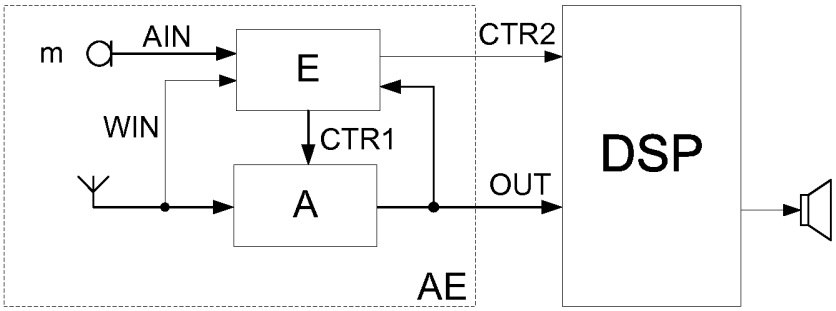

FIG. 7b

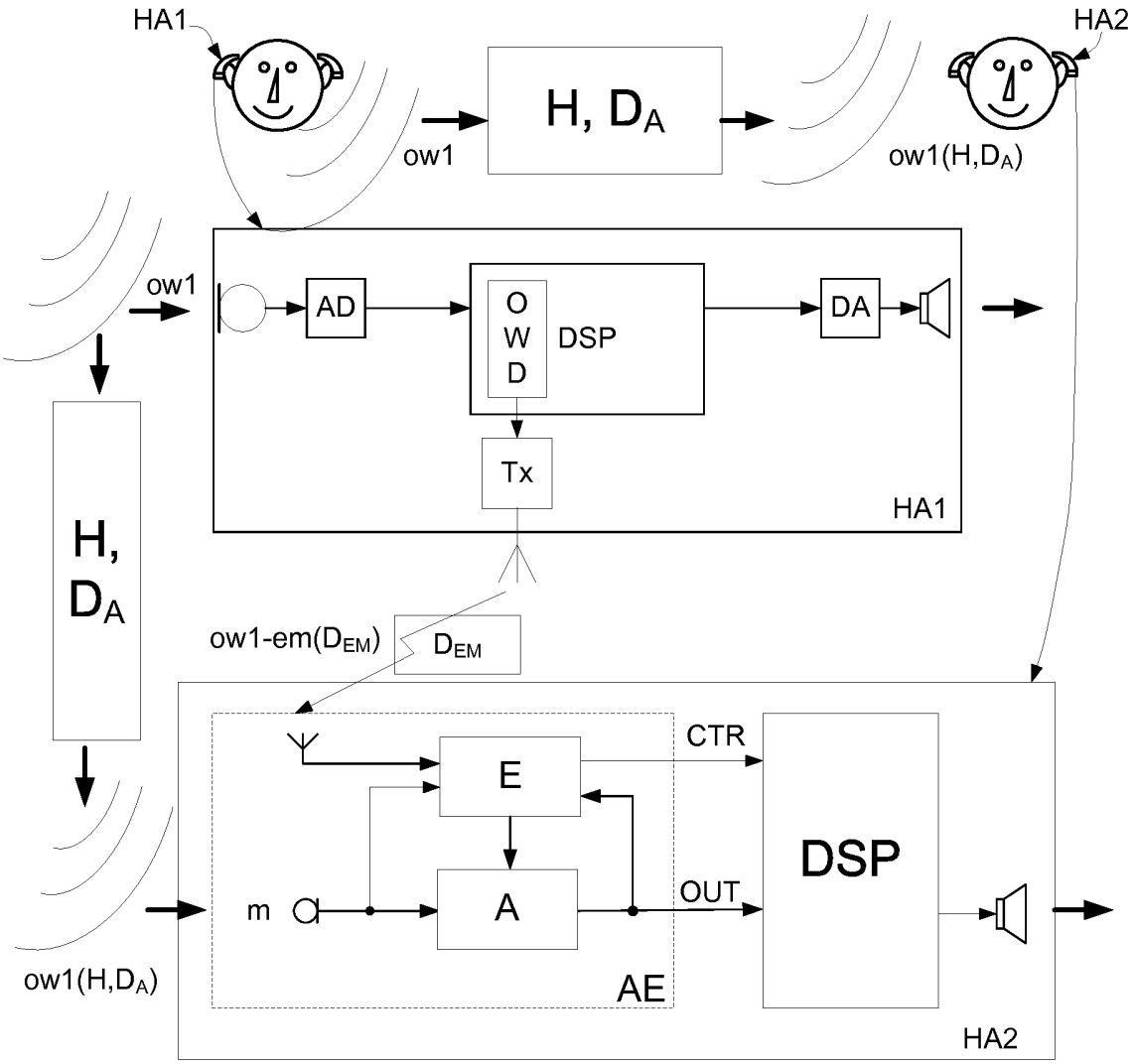
HA1
HA2
H, D_A
ow1
ow1(H,D_A)
ow1
AD
O
W
D
DSP
DA
Tx
HA1
H,
D_A
ow1-em(D_{EM})
D_{EM}
CTR
E
DSP
m
A
OUT
ow1(H,D_A)
AE
HA2

FIG. 8

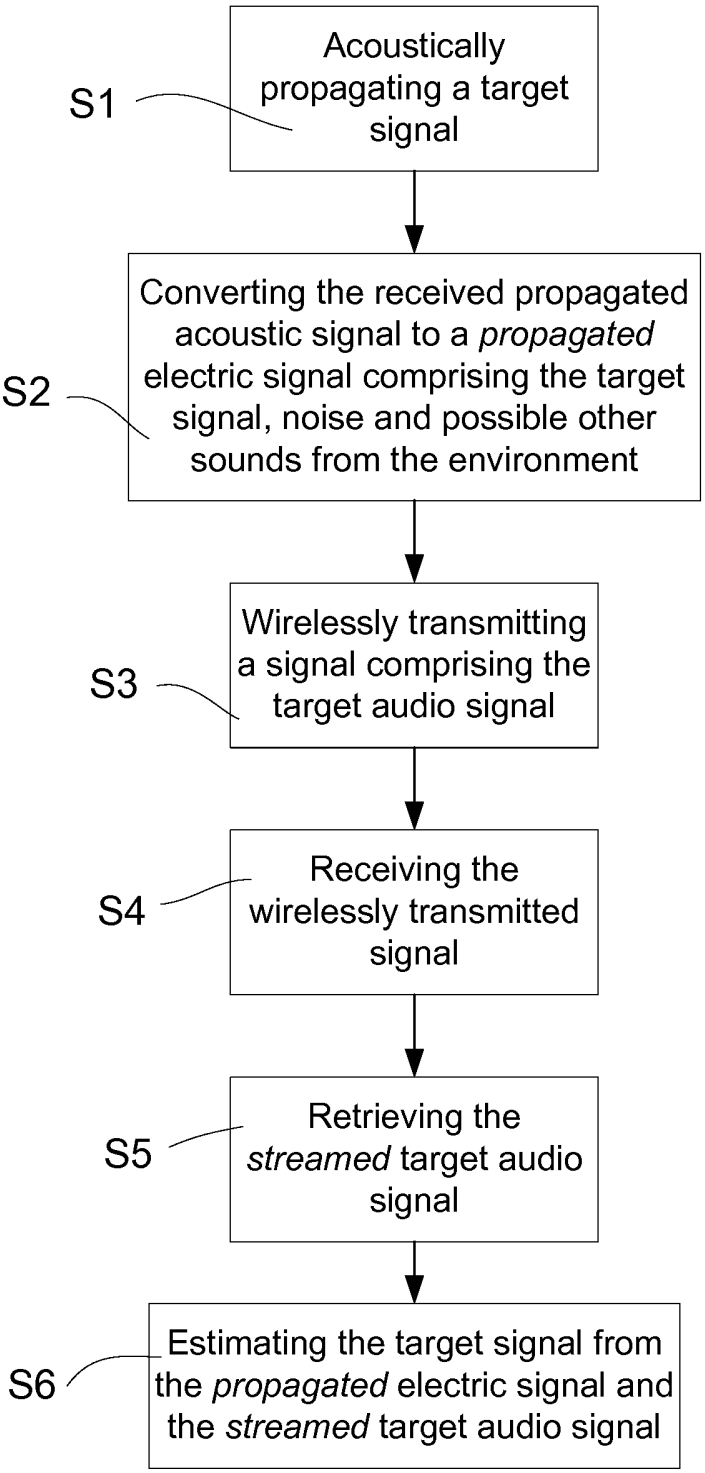
S1
Acoustically propagating a target signal
S2
Converting the received propagated acoustic signal to a *propagated* electric signal comprising the target signal, noise and possible other sounds from the environment
S3
Wirelessly transmitting a signal comprising the target audio signal
S4
Receiving the wirelessly transmitted signal
S5
Retrieving the *streamed* target audio signal
S6
Estimating the target signal from the *propagated* electric signal and the *streamed* target audio signal

FIG. 9

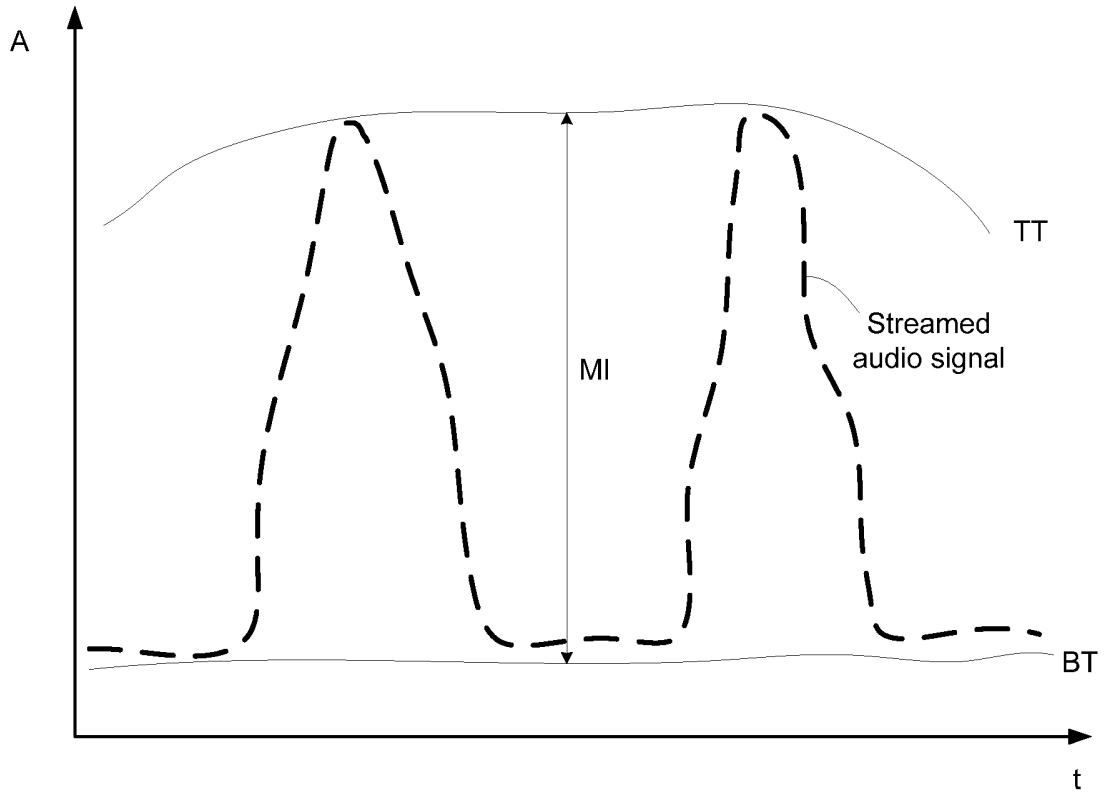

FIG. 10a

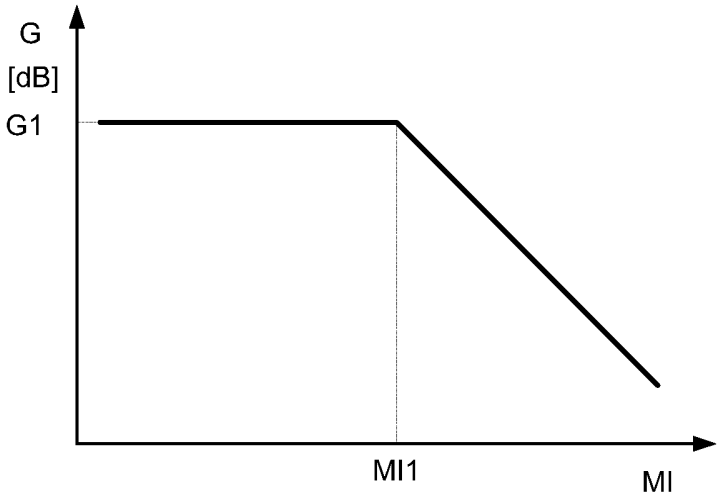

FIG. 10b

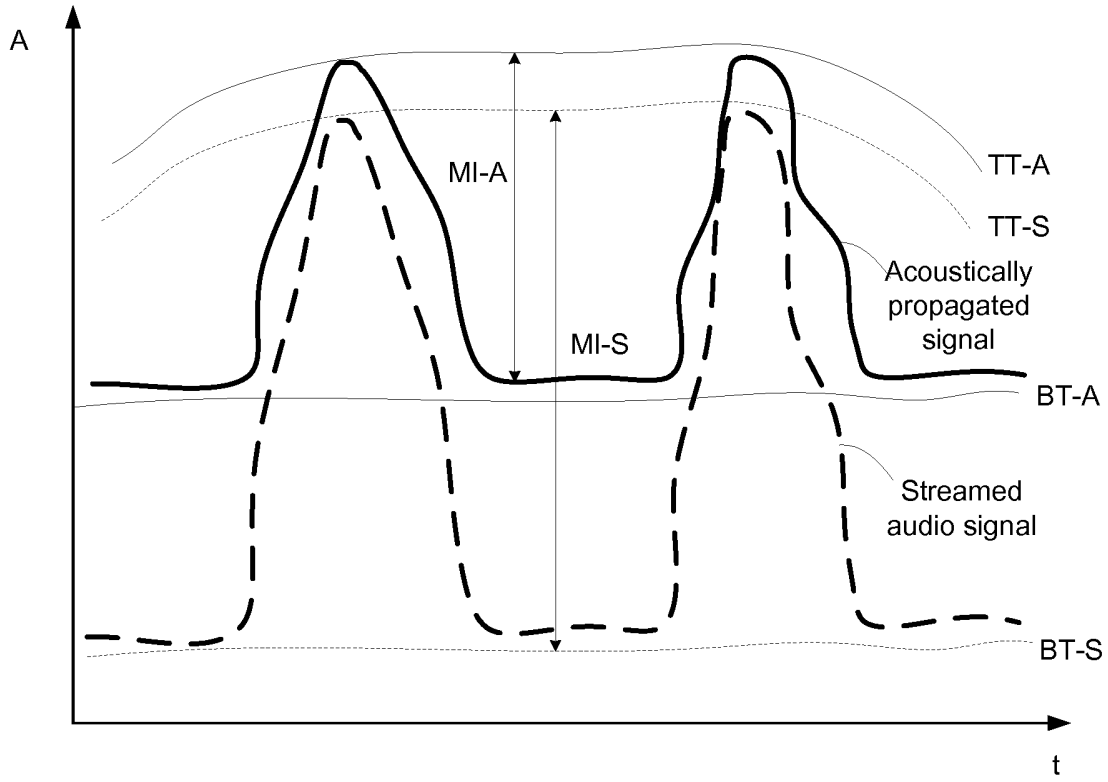

FIG. 11a

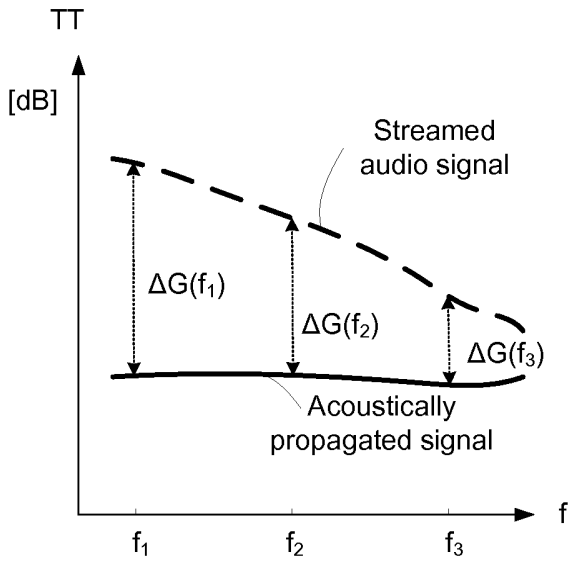

FIG. 11b

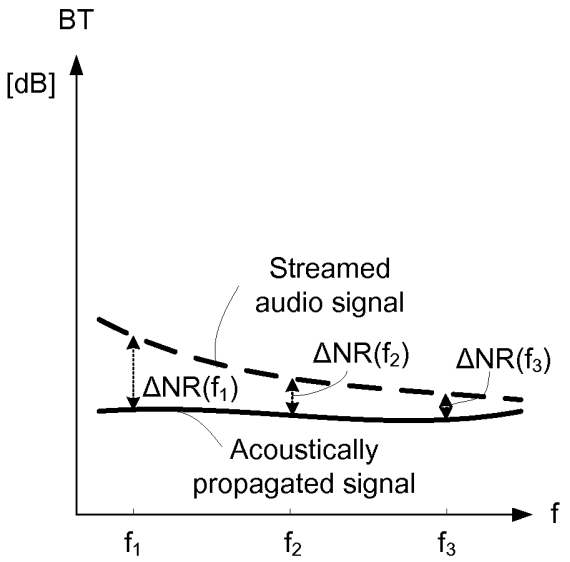

FIG. 11c

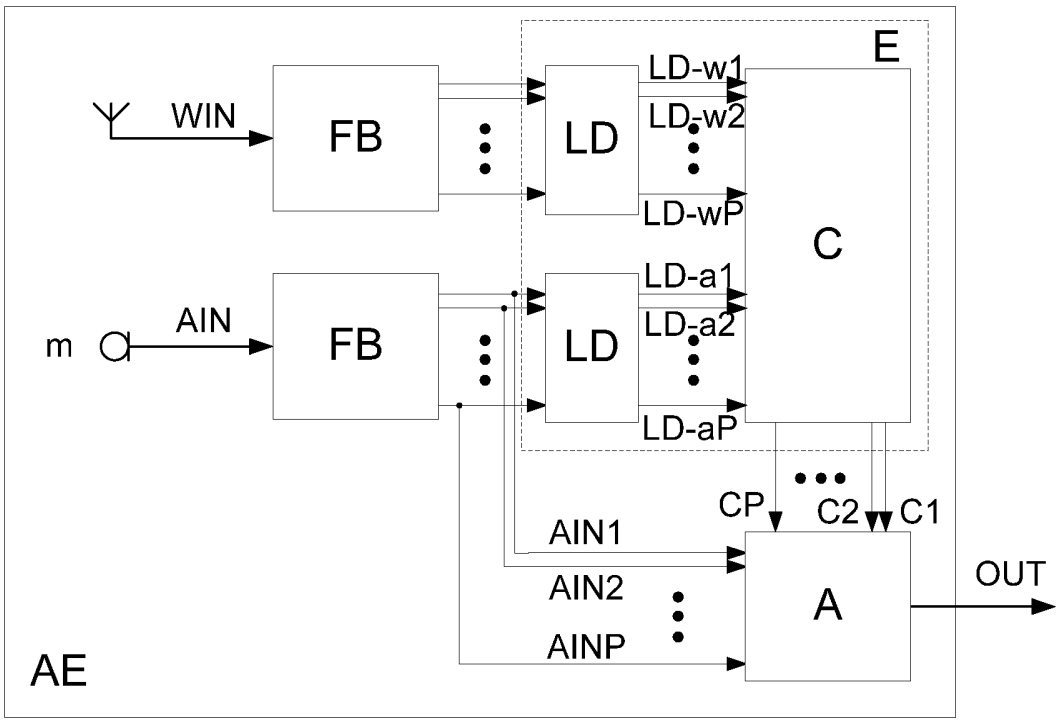

FIG. 12a

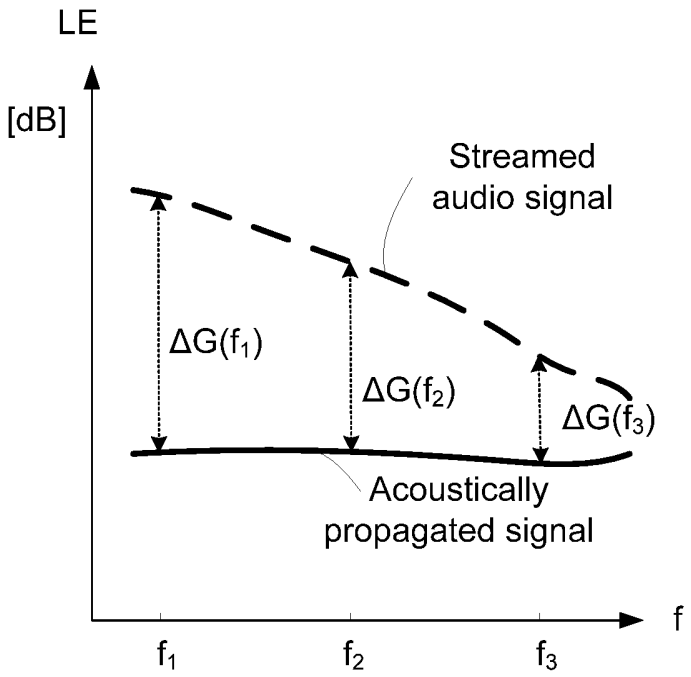

FIG. 12b