The invention relates to a listening device for processing an input sound to an output sound. The invention further relates to a method of estimating a feedback transfer function in a listening device. The object of the present invention is to provide an alternative scheme for minimizing feedback in listening devices. The problem is solved in that the listening device comprises an input transducer for converting an input sound to an electric input signal and defining an input side, an output transducer for converting a processed electric output signal to an output sound and defining an output side, a forward path being defined between the input transducer and the output transducer, and comprising a signal processing unit adapted for processing an SPU-input signal originating from the electric input signal and to provide a processed SPU-output signal, and an electric feedback loop from the output side to the input side comprising a feedback path estimation unit for estimating an acoustic feedback transfer function from the output transducer to the input transducer, and an enhancement unit for estimating noise signal components in the electric signal of the forward path and providing a noise signal estimate output, wherein the feedback path estimation unit is adapted to use the noise signal estimate output in the estimation of the acoustic feedback transfer function. This has the advantage of providing an adaptive feedback cancellation system which is robust in situations with a high degree of correlation between the output signal and the input signal of a listening device. The invention may e.g. be used in hearing aids, head sets, mobile phones, wearable/portable communication devices, etc.
ADAPTIVE FEEDBACK CANCELLATION BASED ON INSERTED AND/OR INTRINSIC CHARACTERISTICS AND MATCHED RETRIEVAL

TECHNICAL FIELD

The present invention relates to methods of feedback cancellation in listening devices, e.g. hearing aids. The invention relates specifically to a listening device for processing an input sound to an output sound. The invention furthermore relates to a method of estimating a feedback transfer function in a listening device. The invention further relates to a data processing system and to a computer readable medium.

The invention may e.g. be useful in applications such as hearing aids, headsets, mobile phones, wearable/portable communication devices, etc.

BACKGROUND ART

The following account of the prior art relates to one of the areas of application of the present invention, hearing aids.

It is well-known that in standard adaptive feedback cancellation systems, correlation between the receiver signal and the microphone target signal, the co-called autocorrelation (AC) problem, leads to a biased estimate of the feedback transfer function. This, in turn, leads to cancellation of (parts of) the target signal and/or sub-oscillation/howls due to bias in the estimate of the feedback transfer function. One way to deal with the AC problem is to rely on AC detectors and decrease convergence rate in sub-bands where AC is dominant, see e.g. WO 2007/1 13282 A1 (Widex). Although this is definitely better than not dealing with the AC problem at all, the disadvantage is that adaptation can be very slow in frequency regions often dominated by AC, e.g. low-frequency regions in speech signals. Another way to deal with the AC problem is to introduce so-called probe noise, where an, ideally inaudible, noise sequence is combined with the receiver signal before play back (being presented to a user). In principle, this well-known class of
methods, see e.g. EP 0 415 677 A2 (GN Danavox), completely eliminates the AC problem. However, since in general the probe noise variance must be very small for the noise to be inaudible, the resulting adaptive system becomes very slow. An improvement can be obtained by using masked noise as e.g. described in US 2007/1 72080 A1 (Philips).

WO 2007/1 251 32 A2 (Phonak) describes a method for cancelling or preventing feedback. The method comprises the steps of estimating an external transfer function of an external feedback path defined by sound travelling from the receiver to the microphone, estimating the input signal having no feedback components of the external feedback path using an auxiliary signal, which does not comprise feedback components of the external feedback path, and using the estimated input signal for estimating the external transfer function of the external feedback path.

Traditional probe noise solution:
Prior art probe noise based solutions of an adaptive feedback cancellation (FBC) system, where, ideally, a perceptually undetectable noise sequence is added to the receiver signal, can in principle completely by-pass the AC-problem. FIG. 1a shows an example of a listening device comprising a traditional adaptive system based on probe noise, where the goal is to approximate the unknown, time-varying transfer function \( F(z,n) \) (representing leakage feedback from receiver to microphone) by an estimate \( Fh(z,n) \), which here is assumed to be an FIR system. A forward path is defined between the microphone and the receiver. The estimate \( Fh(z,n) \) may be updated using any of the standard adaptive filtering algorithms such as NLMS, RLS, etc. (cf. Algorithm unit feeding update filter coefficients to variable filter part \( Fh(z,n) \) in FIG. 1a). The probe noise (generated by Probe signal unit in FIG. 1a) is denoted as \( u(n) \) and can be generated in a variety of ways (cf. e.g. methods A and B discussed below or any other appropriate method, e.g. by filtering a white noise sequence through an analysis-modification-synthesis filter bank, or through an MR filter). The probe signal \( u(n) \) is connected to the Algorithm part of the adaptive FBC-filter as well as being added to output signal \( y(n) \) from the forward gain unit \( G(z,n) \) in output SUM unit ‘+’, whose output \( u(n) \) is connected to the receiver and to the variable filter part \( Fh(z,n) \) of the adaptive FBC-filter. The Algorithm part
additionally bases the estimate of filter coefficients of the variable filter part \( Fh(z,n) \) of the FBC-filter on the feedback corrected input signal \( e(n) \) generated by a subtraction in input SUM unit '+' of the feedback estimate \( vh(n) \) of the variable filter part \( Fh(z,n) \) of the FBC-filter from the input signal comprising feedback signal \( v(n) \) and target signal \( x(n) \) as picked up by the microphone. Due to the preferably inaudible nature of the probe noise, such prior art solutions lead to relatively slow adaption rates of the adaptive system.

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**DISCLOSURE OF INVENTION**

The present invention relates in general to methods for feedback cancellation in listening devices, e.g. hearing aids. The methods can in principle be used with any Dynamic Feedback Cancellation (DFC) system based on the traditional setup where a model (e.g. a FIR or \( MR \) model) of the feedback channel transfer function is updated using any adaptive filter algorithm, e.g. normalized least mean square (NLMS), recursive least squares (RLS), affine projection type of algorithms, etc., see e.g. [Haykin, 1996] or [Sayed, 2003]. While the presented methods are expected to be used in a sub band based system, the concepts are in principle general and may be used in full band based systems as well. Also warping, e.g. in the form of warped filters, cf. e.g. [Harma et al., 2000], may be used in combination with other functional elements (e.g. linear filters, such as FIR or \( MR \) filters) of the present invention. In preferred embodiments, some of, such as a majority of, the features of the present invention are implemented as software algorithms adapted for running on a processor of a portable device, e.g. a communication device or a listening device.

30 In an aspect, the invention relates to the introduction and/or identification of specific characteristic properties in an output signal of the forward path of a listening device. A signal comprising the identified or introduced properties is propagated through the feedback path from output to input transducer and extracted or enhanced on the input side in an Enhancement unit matching (in agreement between the involved units) the introduced and/or identified specific characteristic properties. The signals comprising the specific
characteristic properties on the input and output sides, respectively, (i.e. before and after having propagated through the feedback path) are used to estimate the feedback path transfer function in a feedback estimation unit.

5 Enhancement of characteristics, noise retrieval (noise enhancement):
The invention relates in particular to the retrieval or enhancement of characteristics (e.g. modulation index, periodicity, correlation time, noise or noise-like parts) of a signal in the forward path of a listening device, and to the use of the retrieved or enhanced characteristics in the estimation of acoustic feedback. FIG. 1b illustrates the general concept of and the basic functional elements of a method and system using retrieval or enhancement of characteristics of a signal in the forward path, e.g. intrinsic noise-like signals, in the estimation of the feedback path as suggested by the present invention. The embodiment in FIG. 1b comprises the same elements as the listening device of FIG. 1a, except that the Probe signal generator (in the most general embodiment) is omitted. An Enhancement unit (e.g. a noise retrieval unit) for extracting characteristics (e.g. noise-like parts) of the output signal \( u(n) \) is inserted in a first input path to the algorithm part of the adaptive FBC filter. It takes the output signal \( u(n) \) as an input and provides as an output an estimate \( \text{us}(n) \) consisting of components having certain specified characteristics (e.g. components with a certain modulation index, components with a certain correlation time, e.g. noise-like parts, etc.) of the output signal \( u(n) \), the estimate being connected to the Algorithm part of the adaptive FBC-filter. The ideal purpose of the Enhancement unit is to ensure that the signal \( \text{us}(n) \) is uncorrelated with the (target) input signal \( x(n) \). This may (ideally) e.g. be achieved by filtering out (retrieving) signal components from the receiver signal \( u(n) \), which are uncorrelated with \( x(n) \). Alternatively or additionally, the or an Enhancement unit may be located on the input side of the forward path (cf. the Enhancement unit in FIG. 1b with a dashed outline). In a preferred embodiment, an additional Enhancement unit is provided on the input side (dashed outline in FIG. 1b), which is matched to the Enhancement unit on the output side, in this case to extract the same characteristics from the (here) feedback corrected input signal \( e(n) \) that are extracted or estimated from the output signal \( u(n) \) by the Enhancement unit on the output side.
An object of the present invention is to provide an alternative scheme for minimizing feedback in listening devices.

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

**A listening device:**

An object of the invention is achieved by a listening device for processing an input sound to an output sound. The listening device comprises,

- an input transducer for converting an input sound to an electric input signal and defining an input side,

- an output transducer for converting a processed electric output signal to an output sound and defining an output side,

- a forward path being defined between the input transducer and the output transducer, and comprising a signal processing unit adapted for processing an SPU-input signal originating from the electric input signal and to provide a processed SPU-output signal, and

- an electric feedback loop from the output side to the input side comprising

  - a feedback path estimation unit for estimating an acoustic feedback transfer function from the output transducer to the input transducer, and

  - an enhancement unit for extracting characteristics of an electric signal of the forward path and providing an estimated characteristics output;

wherein the feedback path estimation unit is adapted to use the estimated characteristics output in the estimation of the acoustic feedback transfer function.

This has the advantage of providing an adaptive feedback cancellation system which is robust in situations with a high degree of correlation between the output signal and the input signal of a listening device.

In an embodiment, the output transducer is a receiver (loudspeaker) for converting an electric input (e.g. said processed electric output signal) to an acoustic output (a sound).
The aim of the enhancement unit is to extract signal components with certain pre-specified characteristics (e.g. inserted modulation characteristics, e.g. an AM-function, noise-like signal components, etc.) in the input signal to the enhancement unit, or in other words to eliminate or reduce signal components (in the input to the feedback path estimation unit), which are NOT related to a deliberately inserted probe signal or NOT related to the 'noise' intrinsically present in the signal (e.g. the receiver signal).

The term Originating from' is in the present context taken to mean being equal to or related to by means of attenuation, amplification, compression, filtering or other audio processing algorithms.

In the present context, terms 'noise' or 'noise-like components' in relation to signal components of the listening device (e.g. related to a signal of the forward path, e.g. to an input signal to a receiver of the listening device), refer to signals or signal components (e.g. viewed in a particular frequency range or band), which are uncorrelated with the (target) input signal x(n). This noise or these noise-like components of a signal, typically having very little structure (or short correlation time) and therefore noisy in appearance, is/are of key importance to the present invention.

In the present context, a 'noise like part of the (receiver) signal' is taken to mean one or more components in the (receiver) signal, which are substantially uncorrelated with the input signal. The terms 'uncorrelated' or 'substantially uncorrelated' are in the present context taken to mean 'having a correlation time smaller than or equal to a predefined value'. Since, typically, the receiver signal is approximately a delayed (and scaled) version of the input signal, this is equivalent to saying that a noise-like part of the receiver signal comprises signal components in the receiver signal with a correlation time smaller than the delay of the forward path. For a noise-free speech signal, for example, these components would correspond to time-frequency regions corresponding to 'noise-like' speech sounds such as /s/ and /l/, or high-frequency regions of some vowel speech sounds. For a speech signal contaminated by acoustical noise, these components would typically include time-frequency regions where the acoustical noise is
dominant as well, assuming that the acoustical noise has low correlation
time itself; this is the case for many noise sources, see e.g. [Lotter, 2005].

The term 'time-frequency region' implies that a signal is available in a time-
frequency representation, where a time representation of the signal exist for
the frequency bands constituting the frequency range considered in the
processing. A 'time-frequency region' may comprise one or more frequency
bands and one or more time units. Alternatively, the signal may be available
in successive time units (frames \( F_m \), \( m=1, 2, ... \)), each comprising a
frequency spectrum of the signal in the corresponding time unit (m), a time-
frequency tile or unit comprising a (generally complex) value of the signal in
a particular time (m) and frequency (p) unit. A 'time-frequency region' may
comprise one or more time-frequency units.

The methods of the present invention may in general be used in a full band
processing system (i.e. a system wherein each processing step is applied to
the full frequency range considered). Preferably, however, the full range
considered by the listening device (i.e. a part of the human audible
frequency range (20 Hz - 20 kHz), such as e.g. the range from 20 Hz to 12
kHz) is split into a number of frequency bands (e.g. 2 or more, such as e.g. 8
or 64 or 256 or 512 or 1024 or more), where at least some of the bands are
processed individually in at least some of the processing steps.

In an embodiment, the feedback path estimation unit comprises an adaptive
filter. In a particular embodiment, the adaptive filter comprises a variable
filter part and an algorithm part, e.g. an LMS or an RLS algorithm, for
updating filter coefficients of the variable filter part, the algorithm part being
adapted to base the update at least partly on said noise signal estimate
output from the enhancement unit and/or on a probe signal from a probe
signal generator.

In an embodiment, the input side of the forward path of the listening 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 \( t_n = n \times (1/f_s) \), \( n \) is a sample index, e.g.
an integer \( n = 1, 2, \ldots \) indicating a sample number. The duration in time of \( X \) samples is thus given by \( X/f_s \).

In an embodiment, the signal processing unit is adapted for processing the SPU-input signal originating from the electric input signal in frequency bands.

In an embodiment, the consecutive samples \( S_n \) are arranged in time frames \( F_m \), each time frame comprising a predefined number \( Q \) of digital time samples \( S_q (q = 1, 2, \ldots, Q) \), corresponding to a frame length in time of \( L = Q/f_s \), where \( f_s \) is a sampling frequency of an analog to digital conversion unit (each time sample comprising a digitized value \( S_n \) (or \( s(n) \)) of the amplitude of the signal at a given sampling time \( t_n \) (or \( n \)). A frame can in principle be of any length in time. Typically consecutive frames are of equal length in time. In the present context, a time frame is typically of the order of ms, e.g. more than 3 ms (corresponding to 64 samples at \( f_s = 20 \) kHz). In an embodiment, a time frame has a length in time of at least 8 ms, such as at least 24 ms, such as at least 50 ms, such as at least 80 ms. The sampling frequency can in general be any frequency appropriate for the application (considering e.g. power consumption and bandwidth). In an embodiment, the sampling frequency \( f_s \) of an analog to digital conversion unit is larger than 1 kHz, such as larger than 4 kHz, such as larger than 8 kHz, such as larger than 16 kHz, e.g. 20 kHz, such as larger than 24 kHz, such as larger than 32 kHz. In an embodiment, the sampling frequency is in the range between 1 kHz and 64 kHz. In an embodiment, time frames of the input signal are processed to a time-frequency representation by transforming the time frames on a frame by frame basis to provide corresponding spectra of frequency samples (\( p = 1, 2, \ldots, P \), e.g. by a Fourier transform algorithm), the time-frequency representation being constituted by TF-units (\( m, p \)) each comprising a complex value (magnitude and phase) of the input signal at a particular unit in time (\( m \)) and frequency (\( p \)). The frequency samples in a given time unit (\( m \)) may be arranged in bands \( FB_k (k = 1, 2, \ldots, K) \), each band comprising one or more frequency units (frequency samples).
Noise retrieval. No probe signal inserted (cf. FIG. 1b and 2c. method C):

In an embodiment, no probe signal generator is included in the listening device. In that case the enhancement unit (block Retrieval of intrinsic noise in FIG. 2c) is adapted to extract noise-like parts of the receiver signal (and/or of a signal on the input side), e.g. originating from a speech signal, and to use the extracted noise estimate as an input to the estimation of the acoustic feedback path.

Noise retrieval without inserted probe signal. Processing of signal y(n) on output side and/or signal e(n) on the input side:

In an embodiment, the enhancement unit is adapted for retrieving intrinsic noise-like signal components in the electric signal of the forward path. In a particular embodiment, the enhancement unit is adapted for extracting noise-like parts of the output signal u(n). The enhancement unit takes the output signal u(n) as an input and provides as an output an estimate us(n) of the noise-like parts of the output signal u(n), the estimate being connected to the feedback path estimation unit, e.g. the Algorithm part of an adaptive FBC-filter (cf. e.g. FIG. 1b). Additionally (or alternatively), an enhancement unit for extracting noise-like parts of the feedback corrected input signal e(n) may be inserted (as indicated in FIG. 1b by the dashed outline of the Enhancement unit in the input path for the Algorithm part). The output from the additional or alternative enhancement unit provides an estimate es(n) of characteristics (e.g. noise-like parts) in the feedback corrected input signal e(n), which is connected to the feedback path estimation unit, e.g. the Algorithm part of an adaptive FBC-filter and used in the calculation of update filter coefficients of the variable filter part Fh(z,n) of the adaptive FBC-filter (cf. e.g. FIG. 1b).

The retrieval of intrinsic noise may be combined with insertion of probe signal(s). Examples thereof are described in the section on 'Modes for carrying out the invention' (cf. e.g. FIG. 2e, 2f, 2g, 6b).

In an embodiment, the correlation time N, of the noise signal estimate output from the enhancement unit is adapted to obey the relation N^dG+dA, where
$dG$ is the delay of the forward path and $dA$ is the average acoustic propagation delay of an acoustic sound from the output of the receiver to the input of the microphone, when following a direct physical path (not including reflections e.g. from external objects). In an embodiment, the correlation time $N_i$ of the noise signal estimate output obeys $N_i \leq dG$. The delay of the forward path is in the present context taken to mean the delay from the microphone input via the electric forward path to the output of the receiver. The forward path delay can e.g. be determined by adding the delays of the components constituting the forward path, which are usually known, or measuring the delay acoustically/electrically by applying a known input signal and measuring the resulting output from the receiver. An analysis of the input and output signal allows determining the delay. The average acoustic propagation delay can e.g. be determined in a similar manner with the hearing device mounted on/in the ear.

In an embodiment, the enhancement unit comprises an adaptive filter. In a preferred embodiment, the enhancement unit comprises an adaptive filter $C(z,n)$ of the form

$$C(z,n) = 1 - DR(z) \times LR(z,n)$$

$$= \left\langle z^{N_i} \times \sum_{p=0}^{P} c_p z^{-p} \right\rangle$$

$$= \left\langle \sum_{p=N_i}^{N_i+P} c_p z^{-p} \right\rangle$$

where $C(z,n)$ represents the resulting filter, $DR(z) = z^{N_f}$ represents a delay corresponding to $N_f$ samples, $LR(z,n)$ represents the variable filter part, $N_f$ is the maximum correlation time, and $c_p$ are the filter coefficients adapted to minimize a statistical deviation measure of $us(n)$ (e.g. $\mathcal{E}[\epsilon^2] / \mathcal{E}[us(n)^2]$, where $\mathcal{E}$ is the expected value operator) and $us(n)$ is the noise signal estimate output, and where $P_f$ is the order of $LR(z,n)$. The filter coefficients $c_p$ are estimated here to provide the MSE-optimal linear predictor, although other criteria than MSE (Mean Square Error) may be equally appropriate (e.g. minimize $s\mathcal{E}[us(n)^s]$, where $s > 1$, or any other appropriate statistical deviation procedure). In an embodiment comprising a full band setup, $P_f = 128$ samples (corresponding to 6.4 ms at a sampling rate of 20 kHz). In an embodiment comprising a sub-band setup, the sub-band signals are down-
sampled, so that the efficient sample rate is much lower. The time span, e.g. 6.4 ms can be the same, but since the sample rate is usually much lower, the filter order used for each sub-band filter can then be correspondingly lower.

In a particular embodiment, the enhancement unit(s) is/are fully or partially implemented as software algorithms.

**Retrieval of characteristics AND inserted probe signal (FIG. 1c, 1d, 2a, 2b, 2d, 2e, 2f, 2o, 3, 4a, 4b, 5, 6a, 6b):**

In a particular embodiment, the listening device comprises a probe signal generator for generating a probe signal (e.g. embodied in the signal processing unit). In a particular embodiment, the probe signal contributes to the estimation of the feedback transfer function.

In a particular embodiment, the probe signal generator is adapted to provide that the probe signal has predefined characteristics, and wherein the enhancement unit is adapted to provide a signal estimate output based on said characteristics (it is matched to the predefined characteristics). In a particular embodiment, the characteristics of the probe signal are e.g. selected from the group comprising a modulation index, periodicity, correlation time, noise-like signal components and combinations thereof.

In a particular embodiment, the probe signal generator is adapted to provide that the probe signal has a correlation time $N_0 \leq 64$ samples (corresponding to 3.2 ms at a sampling rate of 20 kHz). Typically, the following tradeoff exists: Increasing $N_0$ allows for higher spectral contrast in the noise, and generally more inaudible noise energy can be inserted. With higher $N_0$, however, an enhancement unit located on the input side can retrieve less of the total noise inserted. Fortunately, the performance of the proposed system does not seem to be very sensitive to an "optimal" choice of $N_0$. Generating a noise sequence with a prescribed correlation time can e.g. be done by filtering a white noise sequence through an FIR shaping filter in that case, the correlation time $N_0$ of the generated noise is simply $P+1$, where $P$ denotes the order of the FIR shaping filter.
Preferably, the probe signal \(us(n)\) is adapted to be inaudible when combined with the output signal \(y(n)\) from the forward gain unit. In an embodiment, \(us(n)\) is adapted to provide that \(u(n)=y(n)+us(n)\) is perceptually indistinguishable from \(y(n)\) for the user of the particular listening device.

In an embodiment, the algorithm part of the feedback path estimation unit comprises a step length control block for controlling the step length of the algorithm in a given frequency region, and wherein the step length control block receives a control input from the probe signal generator. The step length control block adjusts the speed at which the adaptive filter estimation algorithm converges (or diverges). Generally speaking, in spectral regions where a relative large amount of noise has been inserted and/or retrieved, the step length control algorithm would typically increase the convergence rate.

In a particular embodiment, the probe signal generator(s) is/are fully or partially implemented as software algorithms.

FIG. 1c illustrates the general concept of the use of retrieval of characteristics (e.g. noise or any other specific property) AND insertion of a probe signal for estimating a feedback transfer function. The embodiment of a listening device according to the invention in FIG. 1c comprises the same components as the listening device of FIG. 1a. Additionally, the embodiment in FIG. 1c comprises an Enhancement unit for extracting characteristics (e.g. noise-like parts) of the feedback corrected input signal \(e(n)\) and providing an estimate \(es(n)\) of such characteristics to the Algorithm part of the adaptive FBC-filter (instead of the feedback-corrected input signal \(e(n)\)) as discussed in connection with FIG. 1b. The Enhancement unit is matched to the characteristics of the inserted probe signal (be the inserted probe signal characterized by its correlation time, its modulation form, its periodicity, or the like). In the embodiment of FIG. 1c, the Probe signal generator unit receives its input from the output \(y(n)\) from the forward gain unit \(G(z,n)\). The Probe signal unit may alternatively (or additionally) receive its input from the input side of the forward path to provide sufficient processing time for the generation of the Probe signal relative to the output signal \(u(n)\). This is illustrated by the dashed arrow connecting the feedback corrected input
signal $e(n)$ to the Probe signal unit. In general, the probe signal may be generated in any appropriate way, e.g. fulfilling the requirements of non-correlation indicated in the following.

5 Noise generation and noise retrieval. Processing of signal $y(n)$ on output side:

In an aspect of the invention, based on the signal $y(n)$ from a forward path gain unit, a signal $us(n)$ for use in feedback estimation, which is substantially uncorrelated with the input signal $x(n)$, is generated. In some cases $us(n)$ consists of a synthetic noise sequence added to $y(n)$, in other cases $us(n)$ consists of filtered noise replacing signal components in $y(n)$, and in still other cases $us(n)$ consists of signal components already present in $y(n)$. To this end, we propose in particular embodiments a combination of one or more probe signal generation and/or enhancement/retrieval methods (as indicated in the embodiment of FIG. 1d by the blocks Probe signals and/or Retrieval of intrinsic noise in combination with Control block). Some appropriate exemplary probe signal generation methods are:

20 A) Methods based on masked added noise (Block Probe signals in FIG. 1d)
B) Methods based on perceptual noise substitution (Block Probe signals in FIG. 1d)

Methods A and B modify the signal $y(n)$ (cf. e.g. FIG. 1d) by adding/substituting filtered noise, whereas the method of intrinsic noise retrieval mentioned above under the heading 'Noise retrieval. No probe signal inserted' (and referred to in the detailed description of embodiments as Method C) does not modify the signal but simply aims at extracting (retrieving) the signal components which are uncorrelated with $x(n)$, and which are intrinsically present in a signal of the forward path (the intrinsic 'noise-like part of the signal'), e.g. signal $u(n)$ in the embodiments of FIG. 1b and 1d.
Masked probe noise (FIG. 2a, 2d, 2e, 2g, 3, 4a, 4b, 5, 6a, 6b):
In a particular embodiment, the probe signal generator is adapted to provide a probe signal based on masked added noise.

In a particular embodiment, the probe signal generator comprises an adaptive filter for filtering a white noise input sequence \( w \), the output of the variable part \( M \) of the adaptive filter forming the masked probe signal, and the variable part \( M \) of the adaptive filter being updated based on a signal from the forward path by an algorithm part comprising a model of the human auditory system. Preferably, the masked probe signal is based on a signal from the output side. Alternatively or additionally, it may be based on a signal from the input side of the forward path.

Preferably, the correlation time \( N_o \) of the masked probe signal \( u_s(n) \) is adapted to not exceed \( dG+df \), where \( dG \), \( df \) denote the forward and feedback path delay, respectively. That is, \( u_s(n) \) is adapted to be uncorrelated with itself, delayed by an amount corresponding to the combined delay of the feedback path and the forward path, i.e., \( Eus(n)us(n-T)=0 \) for \( T>dG+df \).

Insertion of probe signal by perceptual noise substitution (FIG. 2b, 2d, 2f, 2g, 6b):
In a particular embodiment, the probe signal generator is adapted to provide a probe signal based on perceptual noise substitution, PNS.

In a particular embodiment, the probe signal generator comprises a PNS-part located in the forward path, and bases its output on a perceptual noise substitution algorithm (PNS) for substituting one or more spectral regions of its input signal with filtered noise sequences. Preferably, the PNS-part receives an input from the output side of the forward path, i.e. originating from the signal processing unit. Alternatively or additionally, the PNS-part receives an input from the input side of the forward path, e.g. originating from the feedback corrected input signal.

The purpose of the PNS-part is to process the signal \( y(n) \) so as to ensure that the receiver signal \( u(n) \) is uncorrelated to the (target) input signal \( x(n) \),
at least in certain frequency regions (cf. e.g. FIG. 2b). This is achieved by substituting selected spectral regions of the output signal \( y(n) \) of the forward path unit \( G(z,n) \) (cf. FIG. 1d and 2b) and/or of another signal of the forward path (e.g. the feedback corrected input signal \( e(n) \)) with filtered noise sequences and thereby ensure a predefined degree of (un-) correlation in the frequency regions in question.

Several possibilities exist for deciding which frequency regions can preferably be substituted without substantial perceptual consequences. One is to compare the original and the modified signal using a perceptual model and let the model predict the detectability of the modification. Another is to use a masking model as outlined in relation to the discussion of masked noise (Method A) to identify spectral regions of low sensitivity.

**Feedback Noise Retrieval: Processing of signal \( e(n) \) on input side:**

As shown in FIG. 1d, we propose (in an embodiment of the invention) to process the feedback corrected input signal \( e(n) \) in the enhancement unit block *Retrieval of feedback noise* before the signal enters the \( Fh \) filter estimation block of the feedback cancellation (FBC) system (comprising an adaptive filter comprising algorithm part LR filter estimation and variable filter part \( Fh(z,n) \)). The purpose of the *Retrieval of feedback noise* block is the following. The signal \( e(n) \) comprises inserted characteristics, e.g. noise components, or intrinsic noise components (filtered through the feedback channel \( F(z,n) \) and the estimated feedback channel \( Fh(z,n) \)) along with non-noise components, e.g. speech (which typically have much higher energy). Seen from the \( Fh \) filter estimation block of the FBC system, the noise-like components in \( e(n) \) represent the signal of interest, whereas the 'rest' of \( e(n) \) (here) is considered as 'interference'. The adaptive \( Fh \) filter estimation block may operate using \( e(n) \) as an input, as is done in traditional probe noise solutions (cf. e.g. EP 0 415 677 A2), but due to the unfavourable target noise-to-interference ratio (NIR), the adaptation must be very slow, leading to a system which is generally too slow to track real-world feedback paths. It is, however, possible to significantly improve the NIR by processing the signal to retrieve the target noise (here implemented by the enhancement
unit *Retrieval of feedback noise*) and use this 'enhanced noise' signal as an input to the *Fh filter estimation* block of the FBC system.

The algorithms for noise enhancement/retrieval include, but are not limited to:

I) Methods based on long-term prediction (LTP) filtering.
II) Methods based on binaural prediction filtering.

As mentioned above, any method (or combination of methods) of generating noise, including the methods outlined above are intended to be combinable with any method (or combination of methods) for noise enhancement/retrieval including the methods outlined in the following.

In an embodiment, the enhancement unit comprises an adaptive filter.

*Noise retrieval based on long-term prediction filtering (FIG. 4, 6a, 6b):*

In an embodiment, the enhancement unit is adapted to base the signal estimate output on an adaptive long-term prediction, LTP, filter $D(z,n)$ adapted for filtering a feedback corrected input signal on the input side of the forward path to provide a noise signal estimate output comprising noise-like signal components of said feedback corrected input signal.

In an embodiment, the adaptive LTP filter $D$ has a time varying filter characteristic and is of the specific form

$$D(z,n)=1-DE(z)\times LE(z,n)$$

$$=1-z^{-N^2} \times \sum_{p=0}^{P} d_{p+N^2} z^{-p}$$

$$=1-\sum_{p=N^1}^{N^1+N^2} d_{p} z^{-p}$$

where $D(z,n)$ represents the resulting filter, $DE(z) = z^{N^2}$ represents a delay corresponding to $N^2$ samples, $LE(z,n)$ represents the variable filter part, $N^2$ is the maximum correlation time, $d_{p}$ are the filter coefficients adapted to minimize a statistical deviation measure of $es(n)$ (e.g. $\mathbb{E}[\|es(n)\|^2]$), where $\mathbb{E}$ is
the expected value operator), and $P_2$ is the order of the filter $LE(z,n)$, and where $es(n)$ is the output signal of the filter $D(z,n)$, and

$$es(n) = e(n) - \sum_{k=0}^{P_2} d e(n - N 2^k) = e(n) - z(n) ,$$

where $e(n)$ is a feedback-corrected input signal on the input side at time instant $n$ and $z(n)$ can be seen as a linear prediction of $e(n)$ based on past samples of $e(n)$. The filter coefficients $c_i$ are estimated here to provide the MSE-optimal linear predictor, although other criteria than MSE (Mean Square Error) may be equally appropriate (e.g. minimize $\| es(n) \|$², where $s > 1$).

In an embodiment, $N_2$ is larger than 32, such as in the range between 40 and 200 samples for $f_s = 20$ kHz. In a particular embodiment, $N_2$ is larger than or equal to $N_0 + N$, where $N_0$ represents the correlation time of the probe noise sequence, and $N$ represents the efficient length of the feedback path impulse response ($N = c / R, eff$). In the present context, the feedback path delay ($dF$) is taken to mean the time it takes an impulse in the electrical receiver signal $u(n)$ to be registered in the electrical microphone signal. In the present context, the efficient impulse response length ($c / R, eff$) is taken to mean the time span from the impulse is registered in the electrical microphone signal until the final decay of the impulse response. The feedback path delay can e.g. be estimated from the distance from the receiver to the microphone (and the speed of sound), or determined more accurately using acoustical/electrical measurements.

In an embodiment, the order $P_2$ of the LTP-filter is in the range from 16 to 512.

In an embodiment, the enhancement unit comprises a sensitivity function estimation unit. Basically, this unit aims at compensating for the fact that the hearing aid operates in closed-loop in any practical situation, while the feedback path estimation algorithms are designed with an open-loop situation in mind. By taking the sensitivity function into account, the algorithms are brought closer to the situation for which they were designed, and their performance is improved. The estimation of the sensitivity function
has the largest impact on the performance at high loop gains. The sensitivity function is e.g. discussed in [Forsell, 1997].

*Noise retrieval based on binaural prediction filtering (FIG. 5, 6a, 6b):*

In an embodiment, the enhancement unit is adapted to provide a noise signal estimate output based on binaural prediction filtering, wherein an adaptive noise retrieval unit is adapted for filtering a signal $y_c$ from another microphone, e.g. from the input side of the forward path (e.g. a feedback corrected input signal) of a contra-lateral listening device. The use of a signal from another microphone has the advantage that it allows, in principle, more of the introduced noise to be retrieved than with the LTP method described above. This is the case since the proposed filtering is based on *current* signal samples (from an external sensor) rather than *past* samples from the current sensor.

In an embodiment, the adaptive noise retrieval unit has a time varying filter characteristic described by the difference equation

$$e_s(n) = e(n-N) - \sum_{p=0}^{P_3} e_p y_c(n-p),$$

where $y_c(n)$ represents samples from the other microphone, e.g. an external sensor, and

$$LB(z,n) = \sum_{p=0}^{P_3} e_p z^{-p}$$

represents the variable filter part, where $e_p$ are the filter coefficients adapted to minimize a statistical deviation measure of $es(n)$ (e.g. $\|es(n)\|^2$, where $\mathbb{E}$ is the expected value operator) and where, $N_3$ is a delay in samples and $P_3$ is the order of the filter $LB(z,n)$.

In an embodiment, $N_3$ is chosen in the range $0 \leq N_3 \leq 400$ samples (corresponding to 20 ms at a sampling rate of 20 kHz).

In an embodiment, the order $P_3$ of the filter $LB(z,n)$ is in the range from 32 to 1024 or larger than 1024.

*A method of operating a listening device:*
A method of estimating a feedback transfer function in a listening device comprising a feedback estimation system for estimating acoustic feedback is furthermore provided by the present invention. The listening device comprises a forward path between an input transducer and an output transducer and comprising a signal processing unit adapted for processing an SPU-input signal originating from the electric input signal and to provide a processed SPU-output signal $u$, an electric feedback loop from the output side to the input side comprising a feedback path estimation unit for estimating the feedback transfer function from the output transducer to the input transducer, the method comprising extracting characteristics of the electric signal of the forward path and providing an estimated characteristics output; adapting the feedback path estimation unit to use the estimated characteristics output in the estimation of the feedback transfer function.

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

In an embodiment, characteristics of the electric signal of the forward path comprises one or more of the following: modulation index, periodicity, correlation time, noise or noise-like parts.

In an embodiment, extracting characteristics of the electric signal of the forward path comprises estimating signal components in the electric signal of the forward path originating from noise-like signal parts and the estimated characteristics output comprises a noise signal estimate output.

In an embodiment, noise-like signal parts in the forward path are provided in the form of intrinsic noise in the target signal.

In an embodiment, the method further comprises inserting noise-like signal parts in the forward path, e.g. in the form of a probe signal.
A computer-readable medium:

5 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 diskettes, CD-ROM-, DVD-, or hard disk media, or any other machine readable medium, the computer program can also 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:

20 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.

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

30 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**

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 an example of a listening device comprising a traditional adaptive DFC system based on probe noise (FIG. 1a) and overviews of embodiments of a listening device according to the present invention, FIG. 1b illustrating the general concept of retrieval of characteristics of a signal of the forward path (e.g. intrinsic noise-like signal parts) for use in the estimation of the feedback path, FIG. 1c and 1d illustrating various combinations of the use of retrieval of characteristics of a signal of the forward path AND a probe signal in feedback path estimation,

FIG. 2 shows block diagrams of various embodiments of a listening device comprising an adaptive feedback cancellation system based on probe noise or intrinsic noise, one providing adaptive feedback estimation based on masked probe noise (FIG. 2a), one providing adaptive feedback estimation based on perceptual noise substitution, PNS (FIG. 2b), one providing adaptive feedback estimation based on signal decomposition (intrinsic noise retrieval) (FIG. 2c), one providing adaptive feedback estimation based on masked probe noise and perceptual noise substitution (FIG. 2d), one providing adaptive feedback estimation based on signal decomposition and masked probe noise (FIG. 2e), one providing adaptive feedback estimation based on signal decomposition and perceptual noise substitution (FIG. 2f),
and one providing adaptive feedback estimation based on signal
decomposition, masked probe noise and perceptual noise substitution (FIG. 2g),

5 FIG. 3 shows an embodiment of the invention providing adaptive feedback
estimation based on masked probe noise and (feedback) noise retrieval,

FIG. 4 shows an embodiment of the invention providing adaptive feedback
estimation based on masked probe noise and noise retrieval based on Long
10 Term Prediction filtering (LTP) (FIG. 4a) and an embodiment including a
sensitivity remover (FIG. 4b),

FIG. 5 shows an embodiment of the invention providing adaptive feedback
estimation based on masked probe noise and binaural prediction filtering
based feedback noise retrieval, and

15 FIG. 6 shows an embodiment of the invention providing adaptive feedback
estimation based on masked probe noise, binaural prediction filtering based
feedback noise retrieval and LTP based noise retrieval (FIG. 6a) and an
embodiment of the invention providing adaptive feedback estimation based
on signal decomposition (retrieval of 'intrinsic' noise), masked probe noise,
20 perceptual noise substitution, binaural prediction filtering based feedback
noise retrieval and noise retrieval based on LTP (FIG. 6b).

25 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.

30 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.
According to embodiments of the present invention, methods which allow significantly faster convergence while maintaining the advantage of being robust against the autocorrelation (AC) problem are proposed. The following embodiments of the invention are shown as block diagrams of various functional elements of a listening device. In general the functional components can be implemented in hardware or software as the case may be depending on the current application and restrictions. It is, however, understood that most of the functional blocks shown in the drawings - at least in some embodiments - are intended to be implemented as software algorithms. Examples of such blocks are the forward gain block \( G(z,n) \), the adaptive filter blocks (e.g. feedback estimate transfer function \( F_h(z,n) \) and corresponding Algorithm or Filter Estimation blocks for updating filter coefficients of the feedback estimate transfer function), Enhancement/Noise retrieval blocks, and Probe signal generator blocks.

Traditional probe noise solution:
A prior art probe noise based solution of an adaptive feedback cancellation (FBC) system is shown in FIG. 1a and described in the Background art section above.

Noise retrieval (noise enhancement):
FIG. 1b illustrates the general concept of noise retrieval using enhancement of (possibly) intrinsic noise-like signals in the estimation of the feedback path. The embodiment of a listening device according to the invention in FIG. 1b comprises the same components as the listening device of FIG. 1a, except that the Probe signal generator (and the output SUM unit '+') is omitted so that the output signal to the receiver \( u(n) \) is the output of the forward gain unit \( G(z,n) \). A forward path is defined between the microphone and the receiver. An input side of the forward path is defined by the microphone and an output side of the forward path is defined by the receiver. A delimiting functional unit between input and output side of the forward path can e.g. be a block in the forward gain unit \( G(z,n) \) providing a
frequency dependent gain. An *Enhancement* unit for extracting noise-like parts of the output signal *u*(n) is provided. It takes the output signal *u*(n) as an input and provides as an output an estimate *us(n)* of the noise-like parts of the output signal, the estimate being connected to the *Algorithm* part of the adaptive FBC-filter. Additionally (or alternatively), an *Enhancement* unit for extracting noise-like parts (and/or other characteristics) of the feedback corrected input signal *e*(n) may be inserted (as indicated by the dashed outline of the *Enhancement* unit in the input path for the *Algorithm* part). The output from the (optional) additional *Enhancement* unit provides an estimate *es(n)* of the noise-like parts in the feedback corrected input signal *e*(n), which is connected to the *Algorithm* part of the adaptive FBC-filter and used in the calculation of update filter coefficients of the variable filter part *Fh(z,n)* of the adaptive FBC-filter. In an embodiment, the optional *Enhancement* unit on the input side is absent, in which case the input to the *Algorithm* part is the feedback corrected input signal *e*(n). The notation (e.g. *u*(n), *e*(n)) for signals of the listening device indicates a digital representation, which is preferred. It is therefore understood that in such embodiments that are based on a digital representation of signals, the device comprises analogue to digital (A/D) and digital to analogue (D/A) conversion units, where appropriate (e.g. in the forward paths as part of or subsequent to the microphone and prior to the receiver units, respectively). Further, preferred embodiments comprise processing of signals in a time-frequency framework. In such embodiments, the listening device comprises time to time-frequency conversion units and time-frequency to time conversion units, where appropriate (e.g. filter banks and synthesizer units, respectively, or Fourier transform and inverse Fourier transform units/algorithms, respectively, e.g. in the forward paths as part of in connection with the microphone and receiver units, respectively). Also, a directional microphone system (e.g. providing directionally preferred directions of the microphone sensitivity) may form part of the processing of the input signal, before or after the estimate of the feedback path is subtracted. Further, other functional blocks of a listening device may be integrated with those described in connection with the present invention, e.g. systems or components for noise reduction, compression, warping, etc. The notation (e.g. *G(z,n)* and *Fh(z,n)*) in connection with transfer functions, e.g. for filters, implies a preferred time-frequency representation of the signals, *n* being a time parameter and *z*
indicating a z transform \( z = e^{j\omega} \), where \( j \) is the complex unit \( (j^2 = -1) \) and \( \omega = 2\pi f \), where \( f \) is frequency. Various implementations of an \textit{Enhancement} unit are discussed below (noise retrieval methods I, II and C).

5 \textit{Noise retrieval (enhancement) AND probe noise:}

FIG. 1c illustrates the general concept of the use of noise retrieval AND a probe signal. FIG. 1c is described in the \textit{Disclosure of invention} section above. In general, the probe signal may be generated in any appropriate way fulfilling the requirements of non-correlation indicated in the following.

10 For illustration, various implementations of a \textit{Probe signal} unit for generating a probe signal are discussed below (noise generation methods A, B).

FIG. 1d shows a general block diagram of an embodiment of the proposed listening system. An output signal, \( u(n) \), is connected to a receiver for converting an electric input to an acoustic output. The acoustic output leaks back to the microphone through some (unknown) feedback channel \( F(z,n) \). In addition to the (undesired) feedback signal \( v(n) \), the microphone picks up the (desired) target signal \( x(n) \), e.g. a speech signal. After the microphone (and a possible A/D converter and/or possible time->frequency converter, not shown), an estimate \( vh(n) \) of the feedback signal \( v(n) \) is subtracted from the microphone signal to form a feedback compensated signal \( e(n) = x(n) + v(n) - vh(n) \). This signal is connected to a forward path unit \( G(z,n) \), which represents noise suppression, amplification, compression, etc., to form the processed signal \( y(n) \). Normally, this signal would be identical to the receiver output \( u(n) \), but in some embodiments of the proposed system, we introduce a modification of the signal before outputting it (in FIG. 1d represented by the block \textit{Probe signals Addition and/or substitution of Noisy and/or tonal signals}, termed \textit{Probe signals} block in the following). In the block \textit{Fh filter estimation}, an estimate \( Fh(z,n) \) of the feedback channel \( F(z,n) \) is computed. The \textit{Fh filter estimation} block updates the filter estimate \( Fh(z,n) \) across time using any of the well-known adaptive filtering approaches such as (normalized) Least-Mean Square ((N)LMS), recursive least squares (RLS), methods based on affine projections (AP), Kalman filtering, etc. Clearly, if \( Fh(z,n) \) is ‘close to’ the true (unknown) feedback path \( F(z,n) \), the feedback signal \( v(n) \) will largely be eliminated from the feedback compensated signal \( e(n) \) by the feedback estimate signal \( vh(n) \). In contrast
to most standard systems, in some embodiments of the present invention, the output \( y(n) \) of the forward path unit (or as in FIG. 1d, the output \( u(n) \) of the *Probe signals* block) is processed before it enters the *Fh filter estimation* block, cf. *Retrieval of intrinsic noise* block in FIG. 1d providing an estimate of output noise \( us(n) \). Furthermore, in some embodiments of the present invention, the feedback compensated signal \( e(n) \) is processed before it enters the *Fh filter estimation* block, cf. *Retrieval of feedback noise* block in FIG. 1d providing an estimate of input noise \( es(n) \). Consequently, we propose in some embodiments of the invention to introduce some or all of the blocks denoted in FIG. 1d as *Probe signals*, *Retrieval of intrinsic noise*, and *Retrieval of feedback noise*, accompanied by an appropriate *Control* block.

The general purpose of blocks *Probe signals* and/or *Retrieval of intrinsic noise* is to ensure that the signal \( us(n) \) is substantially uncorrelated with the (target) input signal \( x(n) \). This may be achieved by e.g. generating and adding to the output \( y(n) \) of the forward path unit an inaudible noise sequence, which by construction is uncorrelated with \( x(n) \) (*Probe signals* block in FIG. 1d), and/or replacing time-frequency regions in \( y(n) \) with filtered noise whenever this does not lead to audible artefacts (*Probe signals* block in FIG. 1d), and/or filtering out signal components from the receiver signal \( u(n) \), which are uncorrelated with \( x(n) \) (*Retrieval of intrinsic noise* block in FIG. 1d).

The general purpose of the *Retrieval of feedback noise* block is to filter out/retrieve signal components of the feedback corrected input signal \( e(n) \) originating from noise (e.g. from \( us(n) \)). Signal components in \( e(n) \) which do not originate from \( us(n) \) are, seen from the *Fh filter estimation* block, interference, and should ideally be rejected by the *Retrieval of feedback noise* block.

The blocks *Retrieval of intrinsic noise* and *Retrieval of feedback noise* providing the estimates \( us(n) \) and \( es(n) \), respectively, of noise-like signals may receive other inputs than the output \( u(n) \) and the feedback corrected input signal \( e(n) \). In an embodiment, one or both (as in FIG. 1d) of these noise retrieval blocks receive one or more *External signals* as inputs. Such
signals can e.g. be an acoustic signal picked up by another microphone, either in the same hearing aid or elsewhere, e.g. from a contra-lateral hearing aid, an external device, or other external sensors. In FIG. 1d, the Retrieval of intrinsic noise block may receive - in addition to (or instead of) the output signal \( u(n) \) - an input from the Probe signals block. This input can be the noise sequence inserted by the Probe signals block or information describing in which signal regions the noise is inserted. The Retrieval of intrinsic noise block might then operate primarily in signal regions where noise is NOT inserted by the Probe signals generator.

Further, the embodiment of a listening device shown in FIG. 1d comprises a Control block having (one- or two-way) electrical connection to one or more of the blocks \( G(z,n) \), Probe signals Addition and/or substitution of Noisy and/or tonal signals, Retrieval of intrinsic noise, Fh filter estimation and Retrieval of feedback noise. The Control block is e.g. adapted to monitor and adjust the operation of the adaptive filter in the Fh filter estimation block in order to ensure that the loop gain of the system is appropriate. In some cases the feedback path may change quickly (e.g. when a telephone is placed by the ear), and the loop gain will become momentarily high leading to poor signal quality or even howls. In this case, a purpose of the Control block is to adjust the operation of the blocks \( G(z,n) \), Probe signals Addition and/or substitution of Noisy and/or tonal signals, Retrieval of intrinsic noise, Fh filter estimation and Retrieval of feedback noise, in order to extinguish the howl quickly and bring the system loop gain down. More specifically, based on the amount of inserted/intrinsic and/or retrieved noise in a given signal region, the Control block adjusts the adaptation speed of the adaptive filter. If e.g. a signal region has been substituted by filtered noise, the convergence rate (represented by a step length parameter \( \mu \)) can be increased. The Control block may also base its decisions on results from external detector algorithms, e.g. howl detectors, tonality detectors, loop gain estimators, own voice detectors, etc. (represented by External control signals in FIG. 1d), but also on the combined total gain applied in the forward path \( G(z,n) \) (represented by the arrow between the \( G(z,n) \) and Control blocks).
Rather than basing its decision on the amount of noise inserted by e.g. the 
Probe signals Addition and/or substitution of Noisy and/or tonal signal block,
this procedure can also easily be reversed, such that the Control block 
informs the Probe signals Addition and/or substitution of Noisy and/or tonal signal block to insert an appropriate amount of noise in the receiver signal for a given loop gain (as estimated by a loop gain estimator). Furthermore, in high loop gain situations (as estimated by a loop gain estimator), the Control block may inform the $G(z,n)$ block to reduce the gain applied in the forward path, and in this way reduce the total loop gain. An example of such a feedback control system is discussed in WO 2008/151970 A1.

**Noise generation and/or noise retrieval. Processing of signal $v(n)$ on output side:**

To provide a noise signal $us(n)$, which is uncorrelated with the input signal $x(n)$, we propose a combination of one or more methods (as indicated in the embodiment of FIG. 1d by the blocks Probe signals and/or Retrieval of intrinsic noise in combination with Control block):

A) Methods based on masked added noise (Block Probe signals in FIG. 1d)
B) Methods based on perceptual noise substitution (Block Probe signals in FIG. 1d)
C) Methods based on filtering out intrinsic noise in natural signals (Block Retrieval of intrinsic noise in FIG. 1d).

Methods A and B modify the signal $y(n)$ by adding/substituting filtered noise whereas Method C does not modify the signal but simply aims at extracting (retrieving) the signal components which are uncorrelated with the (target) input signal $x(n)$, and which are intrinsically present in the signal $y(n)$ (the 'noise-like part of the signal').

**Generation of masked noise (Method A, FIG. 2a):**

This method is illustrated by the embodiments of a listening device in FIG. 2a (embodiments $\sigma$ and $\beta$). The method aims at adding to the signal $y(n)$ on
the output side of the forward path a noise sequence \( us(n) \) (a sequence with low correlation time), which is uncorrelated with the input signal \( u(n) \). The noise sequence \( us(n) \) may be generated by filtering a white noise sequence \( w(n) \) through an appropriately shaped, time-varying shaping filter \( M(z,n) \) in order to achieve a desired noise spectral shape and level. The filter \( M(z,n) \) is estimated in block Noise shape and level, based on the signal \( y(n) \), cf. embodiment \( \beta \) in FIG. 2a as described below. The shaping filter \( M(z,n) \) may be found through the use of models of the (possibly impaired) human auditory system, more specifically, using any of the many existing masking models, cf. e.g. [ISO/MPEG, 1993], [Johnston, 1988], [Van de Par et al., 2008].

Ideally, the introduced noise sequence \( us(n) \) has the following properties:

1. **P1:** \( us(n) \) is inaudible in the presence of \( y(n) \), that is, \( u(n)=y(n)+us(n) \) is perceptually indistinguishable from \( y(n) \).
2. **P2:** \( us(n) \) is uncorrelated with \( x(n) \), i.e., \( Eus(n)\cdot x(n+k)=0 \) for all \( k \). This makes it in principle possible to completely by-pass the AC-problem.
3. **P3:** The correlation time \( N_0 \) of \( us(n) \) does not exceed \( dG+dF \), where \( dG \), \( dF \) denote the forward and feedback delay, respectively. That is, \( us(n) \) is uncorrelated with itself delayed by an amount corresponding to the combined delay of the feedback path and the forward path, i.e., \( Eus(n)us(n-\tau)=0 \) for \( T>dG+dF \).

Furthermore, dependent on which version of the Retrieval of feedback noise algorithm is used, see FIG. 1d, (details of the different versions of this block are given below), the following additional noise property is preferably obeyed by the noise sequence \( us(n) \).

4. **P4:** The correlation time \( N_0 \) of the noise sequence \( us(n) \) obeys \( N_0<dG+dF \), i.e. a slight strengthening of requirement P3.

In principle, it is possible to generate a probe noise sequence \( us(n) \) with these characteristics. The well-known problem, however, is that the level of the probe noise should preferably be low, e.g. at least 15 dB below \( u(n) \) \( \{y(n)\} \) on average, for requirement P1 to be approximately valid (for normally
hearing persons), but probably quite a bit more for requirements P3 and P4 to be valid in a low-delay setup, like e.g. a hearing aid.

In the embodiment in FIG. 2a denoted \( \sigma \), the processed output signal \( y(n) \) from the forward path unit \( G(z,n) \) (e.g. providing signal processing to compensate for a hearing loss) is connected to the block Masked probe noise for generating a masked noise based on a model of the human auditory system (which is fully or partially implemented in this block or more specifically in block Noise shape and level in embodiment \( \beta \) of FIG. 2a). The masked noise output \( us(n) \) of the block Masked probe noise is connected to the \( Fh \) filter Estimation unit for estimating the feedback path \( F \). The masked noise output \( us(n) \) is further added to the processed output signal \( y(n) \) from the forward path unit \( G(z,n) \) in SUM-unit '+' providing output signal \( u(n) \), which is connected to the output transducer (receiver) and to the variable filter part \( Fh(z,n) \) of the adaptive FBC-filter. The output of the variable filter part \( Fh(z,n) \) providing an estimate \( vh(n) \) of the feedback signal \( v(n) \) is subtracted from the input signal from the microphone in SUM-unit '+' whose output \( e(n) \) is connected to the input of the forward path unit \( G(z,n) \) and to the \( Fh \) filter estimation unit. The error signal \( e(n) \) is ideally equal to the target signal \( x(n) \), which is added to the feedback signal \( v(n) \) in the microphone, so that the input signal from the microphone is equal to \( x(n) + v(n) \) and thus \( e(n) = x(n) + v(n) - vh(n) \). The Control unit is in one- or two-way communication with the forward path unit \( G(z,n) \), the Masked probe noise unit and the \( Fh \) filter estimation unit, e.g. to monitor and adjust the operation of the adaptive filter in the \( Fh \) filter estimation block (e.g. including an adaptation rate).

The embodiment in FIG. 2a denoted \( \beta \) is identical to the embodiment denoted \( \sigma \) as described above, except - as indicated by the dotted rectangle - that the Masked probe noise unit is implemented by shaping filter unit \( M(z,n) \), which is estimated by Noise shape and level unit based on input \( y(n) \) from the forward path unit \( G(z,n) \). The masked noise \( us(n) \) is provided by the shaping filter unit \( M(z,n) \) based on a white noise sequence input \( w(n) \) and filter coefficients as determined by the Noise shape and level unit based on a model of the human auditory system (which is fully or partially implemented in this block). White noise is in the present context taken to
mean a random signal with a substantially flat power spectral density (in the
meaning that the signal contains substantially equal power within a fixed
bandwidth when said fixed bandwidth is moved over the frequency range of
interest, e.g. a part of the human audible frequency range). The white noise
sequence may e.g. be generated using pseudo random techniques, e.g.
using a pseudo-random binary sequence generator (with a large repetition
number \( N_{psr} \), e.g. \( N_{psr} \geq 1000 \) or \( \geq 10000 \)). The Control unit is in one- or two-
way communication with the forward path unit \( G(z,n) \), the Noise shape and
level unit and the \( Fh \) filter Estimation unit (as in embodiment \( \sigma \)).

Noise generation by perceptual noise substitution (Method B, FIG. 2b):

This method is similar in nature to Method A. We propose here another
algorithm, though, called Perceptual Noise Substitution (PNS), for
generating an imperceptible noise sequence, which is uncorrelated with the
input signal \( x(n) \). Like Method A, the algorithm is embodied in block \( \text{Probe signals} \) in FIG. 1d. The algorithm may be seen as a complement (or an
alternative) to the added masked noise solution described above. The
method is illustrated by the embodiments of a listening device shown in FIG.
2b (embodiments \( \sigma \) and \( \beta \)). The general goal is to process the signal \( y(n) \) so
as to ensure that the receiver signal \( u(n) \) is uncorrelated to the (target) input
signal \( x(n) \), at least in certain frequency regions. To achieve this, the idea is
to substitute selected spectral regions of the output signal \( y(n) \) of the
forward path unit \( G(z,n) \) (cf. signal \( y(n) \) in FIG. 1d and 2b) with filtered noise
sequences and thereby ensure a degree of (un-) correlation in the frequency
regions in question. Thus, rather than adding a low-level noise sequence as
with Method A above, we propose here to completely substitute entire time
frequency ranges or tiles of the receiver signal. Denoting by \( \text{ups}(n) \) the
(filtered) noise sequence substituting parts of \( y(n) \) (cf. FIG. 2b), the
requirements to \( \text{ups}(n) \) are identical to those outlined for Method A (cf. P1,
P2, P3, and optionally P4 above).

The advantage of the proposed procedure is that the desired noise-to-signal
ratio in the substituted signal regions is high, much higher than what can
typically be achieved with other probe noise solutions. Obviously, since the
modified receiver input signal \( u(n) \) ideally should be perceptually
indistinguishable (for a particular user) from the original signal \( y(n) \), not all
time-frequency ranges or tiles can be substituted at all times. Several
possibilities exist for deciding which ranges or tiles can be substituted
without perceptual consequences. One is to compare the original and the
modified signal using a perceptual model, e.g. a simplified version of the
model in [Dau et al., 1996], and let the model predict the detectability of the
modification. Another is to use a masking model as in Method A to decide on
spectral regions of low sensitivity. Other, simpler and probably less accurate,
methodologies based on the log-spectral distortion measure (see e.g.
[Loizou, 2007]) could be envisioned.

In the embodiment in FIG. 2b denoted \( \sigma \), the processed output signal \( y(n) \)
from the forward path unit \( G(z,n) \) (e.g. providing signal processing to
compensate for a hearing loss) is connected to the block \( PNS \) for providing
Perceptual Noise Substitution, including substituting selected bands of the
signal \( y(n) \) with filtered noise, to form the output signal \( u(n) \). The selection of
appropriate bands for substitution is controlled by the \( Control \) unit as
indicated above (e.g. based on a perceptual model, masking model, etc.).
The \( Control \) unit is further in communication with the forward path unit \( G(z,n) \)
and also controls the generation of filter coefficients for the variable filter
part \( Fh(z,n) \) by the \( Fh \) filter Estimation unit. The \( Fh \) filter estimation unit
receives its inputs from the output signal \( u(n) \) (receiver input signal
containing imperceptible noise in selected bands) and from the feedback
corrected input signal \( e(n) \), respectively. Apart from that, the embodiment \( \sigma \)
of FIG. 2b comprises the same functional units connected in the same way
as in the embodiment \( \sigma \) of FIG. 2a.

The embodiment in FIG. 2b denoted \( \beta \) is largely identical to the embodiment
denoted \( \sigma \) as described above. In embodiment \( \beta \), however, two outputs of
the PNS unit are shown, a first PNS-output \( upl(n) \) denoted \textit{No substituted
frequency regions} and comprising frequency bands that have been left
unaltered and a second PNS-output \( ups(n) \) denoted \textit{Substituted frequency
regions} and comprising frequency bands comprising substituted frequency
regions that are ideally substantially uncorrelated to the (target) input signal
\( x(n) \). The two output signals \( upl(n) \) and \( ups(n) \) from the \( PNS \) unit are
combined in \( SUM \) unit '+' to provide the output signal \( u(n) \), which is
connected to the receiver and to the variable filter part $Fh(z,n)$ of the adaptive FBC-filter. Both output signals $upl(n)$ and $ups(n)$ from the PNS unit are connected to the $Fh$ filter estimation unit for - together with the feedback corrected input signal $e(n)$ generating filter coefficients for the variable filter part $Fh(z,n)$ (possibly influenced by the Control unit) providing feedback estimate signal $vh(n)$.

Retrieval of intrinsic noise (signal decomposition, Method C, FIG. 2c):

This method is illustrated by the embodiments of a listening device according to the invention shown in FIG. 2c (embodiments $\sigma$ and $\beta$). The method differs from methods A and B in that it does not modify the output signal $y(n)$ from the forward path unit $G(z,n)$ (so $y(n)=u(n)$). Rather, it filters the signal $y(n)$ in order to identify components intrinsically present in $y(n)$ which are uncorrelated with the input signal $x(n)$. The basic idea here is to observe that the signal $y(n)$ is approximately a (scaled) version of the input signal $x(n)$, delayed by $dG$ samples, $dG$ being the delay of the forward path (in units of the sampling time $T_s=1/f_s$). Consequently, components of $y(n)$ with a correlation time shorter than $dG$ are approximately uncorrelated with $x(n)$. Thus, the identified signal components $\{us(n)\}$ of $y(n)$ should preferably obey property P2 discussed above in connection with generation of masked noise:

P2): $us(n)$ is uncorrelated with $x(n)$, i.e., $Eus(n)\cdot x(n+k)=0$ for all $k$,

and additionally:

P5) The correlation time $N_r$ of the extracted sequence $us(n)$ obeys $N_r \leq dG$.

The signal components with low correlation time, i.e. noise or noise-like signal parts, which are intrinsically present in $y(n)$ are extracted and the corresponding signal connected to the $Fh$ filter estimation block (cf. FIG. 2c).

The extraction is performed in the Retrieval of intrinsic noise block of FIG. 2c. The intrinsic noise components are understood to be parts of the signal $y(n)$ which are noisy in character (although, the signal $y(n)$ is not noisy in traditional sense). More specifically, the noise-like signal parts comprising components with low correlation time in (otherwise noise-free) speech signals could be speech sounds like /s/ and /l/. In the case where the signal $y(n)$ is noisy in a traditional sense, e.g. due to acoustical noise in the
environment or due to microphone noise (or to a deliberately inserted probe signal from a probe signal generator), these components would also be extracted by the Retrieval of intrinsic noise block and in that case the output of the block would be a combination of traditional acoustic noise and intrinsic noise in the target signal (and possibly probe noise). The Retrieval of intrinsic noise block can be implemented using an adaptive filter, e.g. an adaptively updated FIR filter with the following z-transform (cf. e.g. FIG. 2c, embodiment β):

\[ C(z,n) = 1 - DR(z) \times LR(z,n) \]

\[ = 1 - z^{-N_f} \times \sum_{p=0}^{P_1} c_p z^{-p} \]

where \( C(z,n) \) represents the resulting filter, \( DR(z) = z^{N_f} \) represents a delay corresponding to \( N_f \) samples, \( LR(z,n) \) represents the variable filter part, \( N_f \) is the maximum correlation time, and \( c_p \) are the filter coefficients, where \( P_1 \) is the order of \( LR(z,n) \).

The filter coefficients \( c_p \) are updated across time in order to minimize the variance of the output, \( us(n) \), i.e. adapted to minimize \( \mathcal{E} \{ |us(n)|^2 \} \), where \( \mathcal{E} \) is the expected value operator. By doing so, components of the input signal having a correlation time longer than \( N_f \) are reduced. Typically, \( N_f \) is chosen as \( N_f = OG \), the delay of the forward path (\( dG \)), preferably including an average acoustic propagation delay from receiver to microphone. The updating of the filter coefficients \( c_p \) may e.g. be performed using any of the well-known adaptive filtering algorithms, including (normalized) LMS, RLS, etc., cf. LR filter estimation unit in FIG. 2c (j8).

In the embodiment in FIG. 2c denoted \( \sigma \), the processed output signal \( y(n) \) from the forward path unit \( G(z,n) \) (providing signal processing) is connected to the enhancement unit Retrieval of intrinsic noise as well as to the receiver (thereby constituting the output (receiver input) signal). The Retrieval of intrinsic noise unit extracts the noise-like part \( us(n) \) of the output signal \( y(n) \), e.g. as indicated above. The noise-like signal \( us(n) \) is connected to the \( Fh \) filter estimation unit, which provides filter coefficients for the variable filter.
part $F_h(z,n)$ estimating the feedback signal $v(n)$. The Control unit is in one- or two-way communication with the forward path unit $G(z,n)$, the Retrieval of (intrinsic) noise unit and the $F_h$ filter estimation unit. Apart from that, the embodiment $\sigma$ of FIG. 2c comprises the same functional units ($G(z,n)$, $F_h(z,n)$, $F(z,n)$, microphone and receiver units) connected in the same way as the embodiment $\sigma$ of FIG. 2a.

The embodiment in FIG. 2c denoted $\beta$ is identical to the embodiment denoted $\sigma$ as described above, except that the enhancement unit Retrieval of intrinsic noise is implemented by a Delay $D_{R}(z)$ unit, an LR filter estimation unit, an $L_{R}(z,n)$ variable filter unit and a SUM unit '+', (as indicated by the dotted rectangle enclosing these units). The filter $C(z,n)$ described above is implemented by the components Delay $D_{R}(z)$, $L_{R}(z,n)$ and SUM unit '+' enclosed by the dashed rectangle and denoted $C(z,n)$. The Delay $D_{R}(z)$ unit receives as an input the output signal $y(n)$ from the forward path unit $G(z,n)$ (which here is equal to the receiver input signal) and provides an output representing a delayed version of the input (e.g. with a delay corresponding to the delay of the forward path unit $G(z,n)$), which is connected to the LR filter estimation unit as well as to the variable filter unit $L_{R}(z,n)$. The output of the variable filter unit $L_{R}(z,n)$ is subtracted from the output signal $y(n)$ from the forward path unit $G(z,n)$ in SUM unit '+', whose output represents the noise-like part $u_s(n)$ of the output signal $y(n)$ predicted based on previous samples of $y(n)$. The noise-like part $u_s(n)$ of the output signal $y(n)$ is connected to the LR filter estimation unit and used in the calculation of filter coefficients for the variable filter unit $L_{R}(z,n)$ as well as to the $F_h$ filter estimation unit of the feedback cancellation system and used in the calculation of filter coefficients for the variable filter unit $F_h(z,n)$. The Control unit is in one- or two-way communication with the forward path unit $G(z,n)$ and the two $(L_{R}$- and $F_h$-) filter estimation units.

Combination of noise generation and noise retrieval methods A, B, C (FIG. 2d, 2e, 2f):

The noise generation or retrieval methods A, B and C may be mutually combined in any appropriate way (and with possible other schemes for generating appropriate noise sequences and possible other schemes for
retrieving noise). In the embodiments shown, noise is typically added to the forward path on the output side (in the examples shown, after the forward path gain unit \( G(z,n) \)). In practice, this need not be the case. The noise generator(s) may insert noise-like signal parts at any appropriate location of the forward path, e.g. on the input side (before the forward path gain unit \( G(z,n) \)) or in the forward path gain unit \( G(z,n) \) or at several different locations of the forward path.

**Masked noise (Method A) and perceptual noise substitution (Method B)**

(Fig. 2d):

Fig. 2d illustrates a model of an embodiment of a listening device, wherein noise generation Method A (masked noise) and B (perceptual noise substitution) are used in combination. In the embodiment of Fig. 2d, the output signal \( y(n) \) of the forward path gain unit \( G(z,n) \) is connected to a PNS unit that (controlled by the Control unit) substitutes selected spectral regions of the output signal \( y(n) \) (e.g. with spectral content comprising noise-like signal components) and provides an output signal \( \text{up}(n) \) that is substantially uncorrelated to the (target) input signal \( x(n) \), at least in certain frequency regions. In the embodiment of Fig. 2d, the output \( \text{up}(n) \) from the PNS unit is represented by two outputs (as also in Fig. 2b), a first PNS-output \( \text{up}(n) \) denoted *No substituted frequency regions* and comprising frequency bands that have been left unaltered and a second PNS-output \( \text{ups}(n) \) denoted *Substituted frequency regions* and comprising frequency bands comprising substituted frequency regions that are ideally substantially uncorrelated to the (target) input signal \( x(n) \). The two output signals \( \text{up}(n) \) and \( \text{ups}(n) \) from the PNS unit are combined in SUM unit ‘+’ to provide the output signal \( \text{up}(n) \). The output signal \( \text{up}(n) \) is connected to a masked noise generator (indicated by dotted rectangle denoted *Masked probe noise*) comprising a *Noise shape and level* unit for estimating the time-varying shaping filter \( M(z,n) \), which filters a white noise sequence \( w(n) \) and provides as an output the masked noise signal \( ms(n) \). The masked noise signal \( ms(n) \) is added to the second output \( \text{ups}(n) \) from the PNS unit in SUM unit ‘+’ whose output \( us(n) \) is used together with the feedback corrected input signal \( e(n) \) as inputs to *Fh filter estimation* unit for generating filter coefficients for the variable filter part \( Fh(z,n) \) for estimating the feedback path. The *Fh filter estimation* unit is in communication with the Control unit, which is also
connected to the Noise shape and level unit, to the forward path gain unit $G(z,n)$ and to the PNS-unit. The masked noise signal $ms(n)$ is further added to the (combined) output signal $up(n)$ from the PNS unit in SUM unit '+' whose output signal $u(n)$ is connected to the receiver and converted to an acoustic signal as well as to the variable filter part $Fh(z,n)$ of the adaptive FBC-filter. The feedback corrected input signal $e(n)$ is further, as in other embodiments, connected to the forward path gain unit $G(z,n)$. The output and input transducers, feedback $F(z,n)$ and feedback estimation $Fh(z,n)$ paths and signals $v(n)$, $vh(n)$ and $x(n)$ have the same meaning as described in connection with other embodiments of the invention (e.g. FIG. 2a).

The masked noise generation method (Method A, FIG. 2a) and the perceptual noise substitution method (Method B, FIG. 2b) and functional units for implementations thereof are further discussed above. Details of masking of noise and perceptual noise substitution are e.g. discussed by [Painter et al., 2000].

**Masked noise (Method A) and extraction of (intrinsic) noise-like parts (Method C) (FIG. 2e):**

FIG. 2e illustrates block diagrams of two embodiments of a listening device according to the invention, wherein noise generation Method A (masked noise) and C (extraction of intrinsic noise-like parts) are used in combination.

In the embodiment σ of FIG. 2e, the output signal $y(n)$ of the forward path gain unit $G(z,n)$ is connected to a masked noise generator (indicated by dotted rectangle denoted Masked probe noise, cf. also FIG. 2a and the discussion above) comprising Noise shape and level unit (controlled by a Control unit) for estimating time-varying shaping filter $M(z,n)$, which filters white noise sequence $w(n)$ and provides as an output the masked noise signal $ms(n)$, which is added to the output signal $y(n)$ of the forward path gain unit in SUM unit '+' to provide output signal $u(n)$, which is connected to the receiver. The output signal $u(n)$ comprising masked noise is connected to an enhancement unit for retrieval of noise-like signal parts from the input signal (indicated by dotted rectangle denoted Retrieval of intrinsic noise, cf. also FIG. 2c and the discussion of Method C above). The unit for retrieval of intrinsic noise-like signal parts comprises a Delay $DR(z)$ unit, an LR Filter
estimation unit, an \( LR(z,n) \) variable filter unit and a SUM unit '+'. The Delay \( DR(z) \) unit receives as an input the output signal \( u(n) \) and provides an output representing a delayed version of \( u(n) \), which is connected to the LR Filter estimation unit as well as to the variable filter unit \( LR(z,n) \). The output of the variable filter unit \( LR(z,n) \) is subtracted from the output signal \( u(n) \) in SUM unit '+', whose output represents the noise-like parts \( us(n) \) (masked as well as intrinsic) of the output \( u(n) \). The noise-like signal \( us(n) \) is connected to the LR Filter estimation unit as well as to the \( Fh \) filter estimation unit of the feedback cancellation system and used in the calculation of filter coefficients for the variable filter units \( LR(z,n) \) and \( Fh(z,n) \), respectively. The Control unit is in one- or two-way communication with the two (LR- and \( Fh-\) ) Filter estimation units, with the Noise shape and level unit of the Masked probe noise generator and with the forward path gain unit \( G(z,n) \). The feedback corrected input signal \( e(n) \) is used as a second input to the \( Fh \) filter estimation unit and is further, as in other embodiments, connected to the forward path gain unit \( G(z,n) \). The output and input transducers, feedback \( F(z,n) \) and feedback estimation \( Fh(z,n) \) paths and signals \( v(n), vh(n) \) and \( x(n) \) have the same meaning as described in connection with other embodiments of the invention (e.g. FIG. 2a).

Embodiment \( \beta \) of FIG. 2e is largely identical to embodiment \( \sigma \) of FIG. 2e. The two embodiments differ in that in embodiment \( \beta \) of FIG. 2e the input to the Retrieval of intrinsic noise unit is the output \( y(n) \) from the forward path gain unit \( G(z,n) \). This means that the noise retrieval unit extracts noise-like parts \( is(n) \) of the output signal \( (y(n)) \) before a (masked) probe signal \( (ms(n)) \) has been added. Consequently, the masked noise signal \( ms(n) \) is added to the output \( is(n) \) of the Retrieval of intrinsic noise unit to provide the resulting noise estimate \( us(n) \), which is connected to the \( Fh \) filter estimation unit (as in embodiment \( \sigma \)). This has the advantage that the Retrieval of intrinsic noise unit does not have to extract the noise-like parts of the signal that originated from the inserted probe noise.

The masked noise generation method (Method A, FIG. 2a) and signal decomposition method comprising extraction of noise-like parts (Method C, FIG. 2c) and functional units for implementations thereof are further discussed above.
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Perceptual noise substitution (Method B) and extraction of (intrinsic) noise-like parts (Method C) (FIG. 2f):

FIG. 2f illustrates a model of an embodiment of a listening device according to the invention, wherein noise generation Method B (perceptual noise substitution) and C (extraction of (intrinsic) noise-like parts) are used in combination. In the embodiment of FIG. 2f, the output signal $y(n)$ of the forward path gain unit $G(z,n)$ is connected to a PNS unit that (controlled by the Control unit) substitutes selected spectral regions of the output signal $y(n)$ and provides a first output signal $upl(n)$ comprising frequency parts that have been left unaltered (output signal No substituted frequency regions in FIG. 2f) and a second output signal $ups(n)$ comprising frequency parts that have been substituted with spectral content comprising noise-like signal components (output signal Substituted frequency regions in FIG. 2f) that are substantially uncorrelated to the (target) input signal $x(n)$. The two output signals from the PNS unit are combined in SUM unit ‘+’ to provide the output signal $u(n)$, which is connected to the receiver and to the variable filter part $Fh(z,n)$ of the adaptive FBC-filter. The output signal $upl(n)$ from the PNS unit comprising frequency ranges that has been left unaltered is connected to an enhancement unit denoted Retrieval of intrinsic noise enclosed by a dotted rectangle in FIG. 2f and comprising a Delay $DR(z)$ unit, an LR filter estimation unit, an LR$(z,n)$ variable filter unit and a SUM unit ‘+’ (cf. FIG. 2c and the discussion of Method C above), which are adapted for estimating the (intrinsic) noise-like parts of the output signal $upl(n)$ from the PNS unit. The output signal $is(n)$ of the Retrieval of intrinsic noise unit (the output of the SUM unit ‘+’ in the dotted rectangle) is connected to a further SUM unit ‘+’ together with the other output signal $ups(n)$ of the PNS unit comprising the frequency parts that have been substituted with spectral content comprising noise-like signal components. The output of this further SUM unit thus represents the estimate $us(n)$ of the noise-like signal parts of the output signal $u(n)$. The estimate $us(n)$ is connected to the $Fh$ filter estimation unit together with the feedback corrected input signal $e(n)$ and used to update the variable filter part $Fh(z,n)$ of the adaptive FBC-filter for estimating the feedback signal $v(n)$. The LR- and $Fh$- filter estimation units can be influenced via the Control unit, which can also influence and/or receive information from forward path gain unit $G(z,n)$ and the PNS unit. The feedback corrected input signal $e(n)$ is further, as in other embodiments,
connected to the forward path gain unit $G(z,n)$. The output and input transducers, feedback $F(z,n)$ and feedback estimation $Fh(z,n)$ paths and signals $v(n)$, $vh(n)$ and $x(n)$ have the same meaning as described in connection with other embodiments of the invention (e.g. FIG. 2a).

The perceptual noise substitution method (Method B, FIG. 2b) and the signal decomposition method comprising extraction of noise-like parts (Method C, FIG. 2c) and functional units for implementations thereof are further discussed above.

**Masked noise (Method A), perceptual noise substitution (Method B) and extraction of (intrinsic) noise-like parts (Method C) (FIG. 2g):**

FIG. 2g illustrates a model of an embodiment of a listening device according to the invention, wherein noise generation Method A (masked noise), Method B (perceptual noise substitution) and noise retrieval Method C (extraction of (intrinsic) noise-like parts) are used in combination. In the embodiment of FIG. 2g, the output signal $y(n)$ of the forward path gain unit $G(z,n)$ is connected to a PNS unit that (controlled by the Control unit) substitutes selected spectral regions of the output signal $y(n)$ and provides a first output signal $upl(n)$ comprising frequency parts that have been left unaltered (output signal *No substituted frequency regions* in FIG. 2g) and a second output signal $ups(n)$ comprising frequency parts that have been substituted with spectral content comprising noise-like signal components (output signal *Substituted frequency regions* in FIG. 2g) providing frequency regions that are substantially uncorrelated to the (target) input signal $x(n)$.

The first and second output signals from the PNS unit are combined in SUM unit '+' and the resulting combined signal $upx(n)$ is connected to a further SUM unit '+' and to a masked noise generator (as indicated by a dotted rectangle denoted *Masked probe noise*, cf. also FIG. 2a and the discussion above) comprising *Noise shape and level* unit (controlled by a Control unit) for estimating time-varying shaping filter $M(z,n)$, which filters white noise sequence $w(n)$ and provides as an output the masked noise signal $ms(n)$, which is added to the combined output signal $upx(n)$ from the PNS unit in further SUM unit '+' to provide output signal $u(n)$, which is connected to the receiver. The *Noise shape and level* unit further receives input signal $y(n)$ from the forward path gain unit $G(z,n)$. The purpose of this is to enable the
Masked probe noise unit to operate on the forward path signal before \((y(n))\) or after \((upx(n)=upl(n)+ups(n))\) perceptual noise substitution (controlled by the Control unit). The Noise shape and level unit may further receive information from the Control unit regarding which bands have been subject to perceptual noise substitution in the PNS unit, which may advantageously influence the generation of masking noise. The masked noise signal output \(ms(n)\) of shaping filter \(M(z,n)\) is further connected to a gain factor unit ‘x’ for applying gain factor \(\alpha\) to the masked noise signal \(ms(n)\). The gain factor \(\alpha\) can in general take on any value between 0 and 1. In a preferred embodiment, \(\alpha\) is equal to 1 or 0, controlled by the Control unit (cf. output \(\alpha\)). The output \(\alpha \cdot ms(n)\) of gain factor unit ‘x’ is added to the output signal \(ups(n)\) from the PNS unit (comprising substituted frequency regions) in SUM unit ‘+’ providing output signal \(upm(n) = \alpha \cdot ms(n) + ups(n)\).

The listening device further comprises an enhancement unit for retrieval of noise-like signal parts from an input signal (enclosed by dotted rectangle denoted Retrieval of intrinsic noise in FIG. 2g, cf. also FIG. 2c and the discussion of Method C above). The embodiment of a unit for retrieval of noise-like signal parts comprises a Delay \(DR(z)\) unit, an LR filter estimation unit, an \(LR(z,n)\) variable filter unit and a SUM unit ‘+’. The Retrieval of intrinsic noise block (and thus Delay \(DR(z)\) unit) receives as an input the output \(ux(n)\) from SUM unit ‘+’ providing signal \((1-\alpha) \cdot u(n) + \alpha \cdot upl(n)\) via two gain factor units ‘x’ applying gain \((1-ct)\) and \(\alpha\) to signals \(u(n)\) and \(upl(n)\), respectively, where the gain factor \(\alpha\) is controlled by the Control unit. The gain factor \(\alpha\) can in general take on any value between 0 and 1. In a preferred embodiment, \(\alpha\) is equal to 1 or 0, controlled by the Control unit (cf. output \(ct\)). The Delay \(DR(z)\) unit provides an output representing a delayed version of the input \(ux(n)\). The delayed output is connected to the LR filter estimation unit as well as to the variable filter unit \(LR(z,n)\). The output of the variable filter unit \(LR(z,n)\) is subtracted from the input signal \(ux(n) = (1-\alpha)\cdot u(n) + ct \cdot upl(n)\) in SUM unit ‘+’, whose output \(is(n)\) represents an estimate of the noise-like part of the input signal \(ux(n)\). The output \(upm(n) = \alpha \cdot ms(n) + ups(n)\) from SUM unit ‘+’ is added to the estimate \(is(n)\) of noise-like parts of the signal \(ux(n)\) in SUM unit ‘+’, whose output represents the resulting noise-like signal \(us(n)\). If \(\sigma=0\), the retrieval of intrinsic noise block operates on the signal in which noise has just been inserted. If, on the other hand,
σ = 1, the retrieval of intrinsic noise block only operates on signal parts which have not already been substituted by noise. In principle, this could be advantageous since there is in general no need to retrieve the noise which has just been inserted. The noise-like signal \( us(n) \) is connected to the \( Fh \) filter estimation unit of the feedback cancellation system and used in the calculation of filter coefficients for the variable filter unit \( Fh(z,n) \). The Control unit is further in one- or two-way communication with forward path gain unit \( G(z,n) \) and the two (LR- and \( Fh- \) Filter Estimation) units. The electrical equivalent of the leakage feedback from output to input transducer \( F(z,n) \) resulting in input signal \( v(n) \) is added to a target signal \( x(n) \) in SUM unit ' + ' representing the microphone. The feedback estimation \( Fh(z,n) \) resulting in feedback signal \( vh(n) \) is subtracted from the combined input \( x(n) + v(n) \) in SUM unit ' + ' whose output, the feedback corrected input signal \( e(n) \), is, as in other embodiments (cf. e.g. FIG. 2a), connected to the forward path gain unit \( G(z,n) \) and to the \( Fh \) filter estimation unit.

The masked noise generation method (Method A, FIG. 2a), the perceptual noise substitution method (B) and the signal decomposition method comprising extraction of noise-like parts (Method C, FIG. 2c) and functional units for implementations thereof are further discussed above.

**Feedback Noise Retrieval: Processing of signal \( e(n) \) on input side:**

The algorithms for noise enhancement/retrieval include, but are not limited to:

i) Methods based on long-term prediction (LTP) filtering.

ii) Methods based on binaural prediction filtering.

As mentioned above, any method (or combination of methods) of generating noise, including the methods outlined above (methods A, B) are intended to be combinable with any method (or combination of methods) for noise enhancement/retrieval including the methods outlined in the following (methods i, ii and C).
**Masked noise (Method A) and noise retrieval (FIG. 3):**

As an example, FIG. 3 shows a combination of noise generation method A (masked noise) with a noise enhancement/retrieval algorithm (*Retrieval of feedback noise* unit in FIG. 3 (cf. e.g. *Enhancement* unit in FIG. 1c), e.g. implementing Method I as outlined below) in a model of a listening device according to the present invention. The model embodiment of FIG. 3 comprises the same elements as the model embodiment β of FIG. 2a. Additionally, the model embodiment of FIG. 3 comprises enhancement unit *Retrieval of feedback noise* for estimating the signal components of the feedback corrected input signal $e(n)$ which originate from the masked noise signal $us(n)$. The output $es(n)$ of the *Retrieval of feedback noise* unit is connected to the *Fh filter estimation* unit for updating the variable filter part $Fh(z,n)$ of the adaptive FBC-filter for estimating the feedback signal $v(n)$. The other input to the *Fh filter estimation* unit is the masked noise signal output $us(n)$ from the filter $M(z,n)$ of the *Masked probe noise* generator. The *Retrieval of feedback noise* unit is in one or two-way communication with a *Control* unit.

**Noise retrieval based on long term prediction (Method I, FIG. 4):**

When using this method, the correlation time of noise signal $us(n)$ preferably does not exceed $N_o$, i.e., during synthesis of $us(n)$, the signal requirements P1—P3(P4) as outlined in the section on generation of masked noise (Method A) above are preferably obeyed.

The components of $e(n)$ which originate from $us(n)$ may be retrieved from the signal $e(n)$ using the observation that the introduced/intrinsic noise in Methods A, B, C has a limited and known, correlation time, say $N_o$. Assuming that the feedback path $F(z,n)$ is (equivalent to) a FIR filter of order $N$, it follows that the correlation time of the noise picked up at the microphone has a correlation time no longer than $N+N_o$. In other words, signal components in $e(n)$ with *longer* correlation time than $N+N_o$ do not originate from the introduced/intrinsic noise sequence $us(n)$. Thus, introducing a filter in the *Retrieval of feedback noise* block of FIG. 1d, which aims at rejecting signal components with a correlation time longer than $N+N_o$ is proposed. Such a filter can be realized using an adaptively updated FIR filter with the following
z-transform (cf. e.g. FIG. 4, dashed rectangle denoted \( D(z,n) \)), where noise retrieval method I (based on long term prediction) is illustrated in combination with noise generation method A (masked noise, see also the corresponding treatment of the output signal \( y(n) \) to generate masked noise signal \( us(n) \) as discussed above in connection with Method A, and as illustrated in FIG. 2a, embodiment \( \beta \):

\[
D(z,n) = 1 - DE(z) \times LE(z,n) \\
= 1 - z^{-N_2} \times \sum_{p=0}^{P_1} d_p z^{-p} \\
= 1 - \sum_{p=N_2}^{N_2+P_2} d_p z^{-p}
\]

where \( D(z,n) \) represents the resulting filter, \( DE(z) = z^{N_2} \) represents a delay corresponding to \( N_2 \) samples, \( LE(z,n) \) represents the variable filter part, \( N_2 \) is the maximum correlation time, \( d_p \) are the filter coefficients adapted to minimize \( E[es(n)]^2 \), where \( E \) is the expected value operator, and \( P_2 \) is the order of the filter \( LE(z,n) \). The dependency of \( d_p \) on the discrete-time index \( n \) has been omitted. The actual values of parameters \( N_2 \) and \( P_2 \) depend on the application in question (sampling rate, frequency range considered, hearing aid style, etc.). For a sampling rate larger than 16 kHz, and full band processing, typically, \( N_2 \geq 32 \), such as \( > 64 \), such as \( > 128 \). The Fourier transform of the filter is found by replacing \( z \) by \( e^{j\omega} \), \( j \) being the complex unit (\( j^2 = -1 \)) and \( \omega \) being equal to \( 2\pi f \), where \( f \) is the normalized frequency.

The updating of the filter coefficients \( d_p \) is performed in \textit{LE filter estimation} unit in FIG. 4 (a, b). The filter coefficients \( d_p \) may be found adaptively, using any standard adaptive algorithm, such as NLMS, as

\[
d_p^* = \arg \min E[(es(n))^2]
\]

where \( es(n) \) is the output signal of the filter \( D(z,n) \), and

\[
es(n) = e(n) - \sum_{l=0}^{P_2} d_l (e(n - N_2 - l)) = e(n) - z(n),
\]

where \( e(n) \) is a feedback-corrected input signal on the input side at time instant \( n \). On the right-hand side, \( z(n) \), can be seen as a prediction of \( e(n) \),
based on signal samples which are at least \( N_2 \) samples old. The filter coefficients \( d_i \) are estimated here to provide the MSE-optimal linear predictor, although other criteria than MSE (Mean Square Error) may be equally appropriate. By doing so, components of the signal \( e(n) \) having a correlation time longer than \( N_2 \) are reduced. \( N_2 \) may preferably be chosen as \( N_2 = N_0 + N \), where \( N_0 \) represents the correlation time of the (probe) noise sequence, and \( N \) represents the delay in the feedback path, in order to reject signal components clearly not originating from the introduced/intrinsic noise. Often, \( D(z,n) \) is called a long-term prediction (LTP) error filter, a term coined in the area of speech coding [Spanias, 1994]. The important thing to note is that the LTP error filter can be considered as a whitening filter, but due to the special structure of \( D(z,n) \) with \( N_2 > 0 \), the output is in general not completely white. In an embodiment, \( N_2 = 0 \) is taken to mean \( N_2 \geq 32 \), such as \( \geq 64 \) or \( \geq 128 \).

By doing so, the NIR may be significantly improved and the adaptation rate of the \( Fh \) filter estimation block can be increased beyond what is possible with traditional systems based on probe noise.

In the proposed setup, the (probe) noise properties and the LTP error filter \( D(z,n) \) are chosen such that their characteristics match: The introduced/intrinsic noise has a correlation time shorter than \( N_0 \), while \( D(z,n) \) reduces signal components with a correlation time longer than \( N_2 = N_0 + N \). In an embodiment, \( N_0 \) is in the range from 32 to 128 samples (assuming a sampling rate of 20 kHz). In this way, \( D(z,n) \) can be seen as a matched filter. If \( N \) is e.g. equal to 64, this leads to \( N_2 \) in the range from 96 to 192. The idea of introducing (probe) noise with certain characteristics (in this case in the autocorrelation domain) is easy to generalize: Alternatively, for example, certain probe signal characteristics in the modulation domain can be introduced and a corresponding matched filter in this domain designed.

In FIG. 4, the adaptive filter \( D(z,n) \) is correspondingly implemented in Retrieval of feedback noise block by units \( \text{Delay } DE(z) \), \( \text{LE}(z,n) \), and SUM '+' (as indicated by the corresponding dashed enclosing rectangle denoted \( D(z,n) \)) providing output \( es(n) \). In the embodiment of FIG. 4a, the Delay \( DE(z) \) unit receives feedback corrected input signal \( e(n) \) as an input and
provides a delayed output which is connected to the algorithm and variable filter parts \textit{LE filter estimation} and \textit{LE(z,n)}, respectively. The output of the variable filter part \textit{LE(z,n)} is subtracted from the input signal \(e(n)\) in SUM unit '+'. The output of the adaptive filter \(D(z,n)\) (i.e. output of \textit{Retrieval of feedback noise} block, i.e. output of SUM-unit '+') in FIG. 4) is the signal \(es(n)\) representing the noise-like part of the (feedback corrected) input signal \(e(n)\). The signal \(es(n)\) is connected to the variable filter part \textit{LE filter estimation} of the adaptive filter \(D(z,n)\) as well as to the \textit{Fh filter estimation} part of the FBC-filter and used in the latter to estimate of filter coefficients for estimating the feedback signal \(v(n)\). The other input to the \textit{Fh filter estimation} unit is the signal \(us(n)\) providing a masked noise signal generated by \textit{Masked probe noise} unit (cf. FIG. 2a) implemented by shaping filter unit \(M(z,n)\), which is estimated by \textit{Noise shape and level} unit based on input \(y(n)\) from the forward path unit \(G(z,n)\). The masked noise \(us(n)\) is provided by the shaping filter unit \(M(z,n)\) based on a white noise sequence input \(w(n)\) and filter coefficients as determined by the \textit{Noise shape and level} unit based on a model of the human auditory system. The masked noise \(us(n)\) is added to the output \(y(n)\) from the forward path unit \(G(z,n)\) in SUM unit '+' to provide output signal \(u(n)\) connected to the receiver and to the variable filter part \(Fh(z,n)\) of the adaptive FBC filter. A \textit{Control} unit is in one- or two-way communication with the forward path gain unit \(G(z,n)\), the \textit{Noise shape and level} unit and the \textit{LE-} and \textit{Fh- filter estimation} units. The electrical equivalent \(F(z,n)\) of the leakage feedback from output to input transducer resulting in input signal \(v(n)\) is added to a target signal \(x(n)\) in SUM unit '+' representing the microphone. The feedback estimation \(Fh(z,n)\) (variable filter part of an adaptive FBC filter) resulting in feedback signal estimate \(vh(n)\) is subtracted from the combined input \(x(n) + v(n)\) in SUM unit '+' whose output, the feedback corrected input signal \(e(n)\), is connected to the forward path gain unit \(G(z,n)\) and (in the embodiment in FIG. 4a) to the \textit{Retrieval of feedback noise} unit (here to the \textit{Delay DE(z)} unit).

The embodiment of a listening device according to the invention shown in FIG. 4b is largely identical to the one shown in FIG. 4a. The differences are the following: In addition to the functional blocks of the embodiment of FIG. 4a, the embodiment of FIG. 4b comprises an \textit{Inv-sensitivity function estimation} block comprising an adaptive filter with an algorithm part \(S\) filter...
estimation and a variable filter part $S(z,n)$ getting its filter coefficient updates from the S filter estimation part. This filter update may be achieved through classical methods such as NLMS. The FIR filter $S(z,n)$ is an estimate of the so-called inverse sensitivity function. The sensitivity function concept in closed-loop identification (see e.g. [Forsell, 1997]) describes the coloration of (intrinsic or introduced) noise components due to the fact that the system is closed-loop. Had the system been open-loop, the sensitivity function would have been $S(z,n)=1$. Strictly speaking, the proposed algorithms for feedback path estimation assume the system to be open-loop, but any hearing aid system is, obviously, closed-loop. By taking into account the sensitivity function, it is possible to bring the situation "experienced" by the $Fh$ filter estimation block closer to being open-loop, and consequently achieve better performance. Specifically, this is done by filtering $e(n)$ in the filter $S(z,n)$ receiving its update filter coefficients from the S filter estimation part of the Inv-sensitivity function estimation block.

**Noise retrieval based on long term prediction (Method I) combined with any noise generation method:**

FIG. 4 illustrates as described above a combination of noise retrieval based on long term prediction (Method I) with noise generation based on the generation of masked noise (Method A). Noise retrieval method I may, however, be combined with any other noise generation method, alone or in combination with other noise generation methods.

Among the advantages provided by embodiments of the present invention with noise retrieval based on LTP are:

- Higher gain possible, especially for tonal signal regions (which are usually considered difficult to handle in traditional systems).
- Significantly reduced distortions in audio signals.
- Fewer howls / distortions as feedback path estimate is generally healthier.
- Proposed algorithm is particularly strong in signal regions with tonal components as these have long correlation times. This is particularly interesting as (any) standard system has weaknesses in such regions.
- Can be used in single HA situations.
Noise retrieval based on binaural prediction filtering (Method II) (FIG. 5):

The general idea in Method I proposed above is to use far-past samples of the error signal \( e(n) \) to predict the current sample of \( e(n) \), and in this way reduce signal components in the error signal estimate \( es(n) \) which are not due to the introduced/intrinsic noise. Clearly, this framework is not dependent of which signal samples are used to predict the current error signal sample \( e(n) \), as long as the signal samples used are uncorrelated with the introduced/intrinsic noise and do correlate to some extent with the current error signal sample. Based on this observation, it is proposed to use signal samples from another microphone, e.g. from a contra-lateral microphone to predict the components of the error signal \( e(n) \), which do not originate from the introduced/intrinsic noise \( us(n) \). The setup is shown in FIG. 5, where a combination of noise retrieval method II based on binaural prediction filtering with noise generation method A based on masked noise is implemented. In an embodiment, non-linearity is introduced into the forward path, e.g. by frequency transposition or PNS. FIG. 5 shows a noise based DFC system using a signal \( y_c(n) \) from another microphone (i.e. e.g. a signal from an external sensor, e.g. a contra-lateral listening device located at another ear than the current one) for retrieving the signal components in \( e(n) \) originating from \( us(n) \). In the embodiment of FIG. 5, the signal \( y_c(n) \) is a processed version of an additional microphone signal (cf. block \( \gamma \), e.g. a feedback corrected microphone signal, as received via a connection to another device (cf. indication Wired or wireless transmission). In FIG. 5, the LTP error filter \( D(z) \) of Method I (cf. FIG. 4) has been replaced by another FIR filter structure (implemented in Binaural retrieval of feedback noise block in FIG. 5), described by the difference equation

\[
es(n) = e(n-N_3)^{-1} - \sum_{p=0}^{P_1} e_p y_c(n-p),
\]

where \( y_c(n) \) represents samples from the external sensor,

\[
LB(z, n) = \sum_{p=0}^{P_1} e_p z^{-p}
\]

represents the variable filter part, where \( e_p \) are the filter coefficients adapted to minimize \( E(es(n)^2) \), where \( E \) is the expected value operator and where \( es(n) \) is the output signal of the proposed filter structure, \( N_3 \) is a delay which may be needed to account for the fact that a latency may be introduced for
transmitting a signal from another sensor to the current one and \( P_3 \) is the order of the filter \( \text{LB}(z,n) \). The purpose of this filter is identical to that of the predictor of \( D(z,n) \) of method 1, namely to predict samples of the error signal \( e(n) \) in order to eliminate signal components NOT related to the probe signal. Specifically, the filter coefficients \( \hat{e}_p \) are found so as to minimize \( E[\hat{e}(n)^2] \). However, in contrast to the predictor of \( D(z,n) \), the predictor \( \text{LB}(z,n) \) bases the prediction, not on \( e(n) \), but on samples from a signal \( \hat{y}_c(n) \) from another (e.g. a contra-lateral) microphone.

Consequently, when using this feedback noise retrieval technique, the introduced/intrinsic noise should preferably have properties P1-P3 (as outlined in the section on generation of masked noise (Method A) above), and in addition preferably:

P6) The introduced/intrinsic noise \( \hat{u}(n) \) is uncorrelated with the contra-lateral microphone signal \( \hat{y}_c(n) \), i.e., \( E\hat{u}(n)\hat{y}_c(n+k) \sim 0 \) for all \( k \).

In FIG. 5, the proposed filter structure is implemented in Binaural retrieval of feedback noise block by units Delay DB(z), LB Filter Estimation, LB(z,n), and SUM '+'. The Delay DB(z) unit receives (feedback corrected) input signal \( e(n) \) as an input and provides a delayed output \( \hat{ed}(n) \) which is connected to SUM unit '+'. The algorithm and variable filter parts LB filter estimation and LB(z,n), respectively, receive input \( \hat{y}_c(n) \) originating from another microphone than the one on which signal \( e(n) \) is based (\( \hat{yc}(n) \) being transmitted by wire or wirelessly, e.g. from a microphone of a contra-lateral device or from another microphone of the same listening device or from another device; the microphone signal from the other microphone has been processed in processing unit Y, e.g. to provide a feedback corrected version of the input signal). The output of the variable filter part LB(z,n) is subtracted from the output signal \( \hat{ed}(n) \) of the Delay DB(z) unit in SUM unit '+'. The output of the filter structure of the Binaural retrieval of feedback noise block (output of SUM-unit '+' in FIG. 5) is the signal \( \hat{es}(n) \) representing the noise-like part of the (feedback corrected) input signal \( e(n) \). This signal \( \{\hat{es}(n)\} \) is connected to the variable filter part LB filter estimation of the filter structure as well as to the Fh filter estimation part of the FBC-filter and in the latter used in the estimate of filter coefficients for estimating the feedback signal \( v(n) \) provided as \( \hat{vh}(n) \) by variable FBC-filter part Fh(z,n). The LB filter
estimation part of the filter structure is electrically connected to a Control unit. The other input to the Fh filter estimation unit is the signal \( \text{usd}(n) \) (an appropriately delayed version of \( \text{us}(n) \) delayed in Delay DB(\( z \)) unit, equal to the other delay unit of the Binaural retrieval of feedback noise block). Signal \( \text{us}(n) \) is a masked noise signal generated by Masked probe noise unit (cf. FIG. 2a) implemented by shaping filter unit \( M(z,n) \), which is estimated by Noise shape and level unit based on input \( y(n) \) from the forward path unit \( G(z,n) \). The masked noise \( \text{us}(n) \) is provided by the shaping filter unit \( M(z,n) \) based on a white noise sequence input \( w(n) \) and filter coefficients as determined by the Noise shape and level unit based on a model of the human auditory system. A Control unit is in one- or two-way communication with the Noise shape and level unit and the LB- and Fh- filter estimation units and the forward path gain unit \( G(z,n) \). The masked noise \( \text{us}(n) \) is added to the output \( y(n) \) from the forward path unit \( G(z,n) \) in SUM unit '+' , the sum providing output signal \( u(n) \) to the receiver. The output signal \( u(n) \) is connected to the variable filter part \( \text{Fh}(z,n) \) of the adaptive FBC-filter. The electrical equivalent \( F(z,n) \) of the leakage feedback from output to input transducer resulting in input signal \( v(n) \) is added to a target signal \( x(n) \) in SUM unit '+' representing the microphone. The feedback estimation \( \text{Fh}(z,n) \) (variable filter part of an adaptive FBC filter) resulting in feedback signal estimate \( \text{vh}(n) \) is subtracted from the combined input signal \( x(n) + v(n) \) in SUM unit '+' whose output, the feedback corrected input signal \( e(n) \), is connected to the forward path gain unit \( G(z,n) \) and to the Binaural retrieval of feedback noise unit, here specifically to the Delay DB(\( z \)) unit. The Binaural retrieval of feedback noise unit is in FIG. 5 represented by units enclosed by the dotted polygon, i.e. including units Delay DB(\( z \)), LB Filter Estimation, DB(\( z,n \)), and SUM '+' as outlined above AND delay unit Delay DB(\( z \)) for delaying masked noise signal \( \text{us}(n) \) to adapt it to the delay of \( \text{es}(n) \) before entering the Fh filter estimation unit.

As mentioned above, the goal of the proposed filter structure is similar to that of \( D(z,n) \) of method I and the coefficients of the proposed filter structure can be estimated and updated in a similar fashion, using e.g. NLMS. However, whereas \( D(z,n) \) is dependent on samples of the microphone signal only (in fact, in the embodiment of FIG. 4a, \( D(z,n) \) is derived from the feedback compensated signal, \( e(n) \)), the proposed filter structure is
dependent on the spatial configuration of sound sources. This is clear from
the observation that $LB(z,n)$ aims at representing the transfer function from
one ear to the other (in case of using a signal originating from a microphone
of a contra-lateral device), which is related to head related transfer functions
HRTF (in the case of a single point source in the free field, this relation is
particularly simple), which in turn are functions of the direction-of-arrival of
the sound source. Further, whereas $D(z,n)$ is dependent on far-past samples
of the error signal, the proposed filter structure may potentially be based on
current samples of the contra-lateral microphone signal. This would be
reflected by choosing $N_3=0$.

Noise retrieval based on binaural prediction filtering (Method II)
combined with any noise generation method:

FIG. 5 illustrates as described above a combination of noise retrieval method
II based on binaural prediction with noise generation method A based on
masked noise generation. Noise retrieval method II may, however, be
combined with any other noise generation methods, alone or in combination.

Among the advantages provided by embodiments of the noise retrieval
method II of the present invention based on binaural prediction filtering are:
- Higher gain possible without howls/distortions, in principle, for any input
  signal, tonal or not.
- Proposed algorithm is in principle strong for any input signal as long as the
  spatial configuration is simple (not too many reflections) and somewhat
  stationary across time.
- Somewhat complementary to the LTP solution proposed above. The LTP
  solution is signal dependent whereas the proposed solution is signal
  independent but dependent on spatial configuration.

The method requires dual, e.g. contra-lateral, listening devices or another
microphone signal from the same listening device or from another device,
e.g. from a communication device, e.g. from an audio selection device.
Combination of noise retrieval methods I, II and C with noise generation methods A, B (FIG. 4, 5, 6):

In general, combinations of one or more of the noise generation methods A, and B with one or more of the noise retrieval methods I, II and C can advantageously be implemented using at least one algorithm from each class.

Noise retrieval based on long term prediction filtering (Method I) and binaural prediction filtering (Method II) combined with noise generation method based on masked noise (Method A):

FIG. 6a shows a model for an embodiment of a listening device according to the invention, wherein noise generation method A based on masked noise is combined with noise retrieval method I based on long term prediction filtering as well as with noise retrieval method II based on binaural prediction filtering. In FIG. 6a, masked noise $us(n)$ (Method A, cf. above) is inserted in the output part of the forward path by block Masked probe noise and used as a first input to the algorithm part (Fh filter estimation) of the adaptive FBC-filter for estimating the feedback path. The noise in the feedback corrected input signal $e(n)$ originating from the inserted masked noise is retrieved in enhancement unit Retrieval of feedback noise using long term prediction filtering (Method I, filter $D(z,n)$, cf. above) and noise from an alternative (possibly processed) microphone signal $yc(n)$ (e.g. from a contra lateral device) is retrieved in enhancement unit Binaural retrieval of feedback noise using binaural prediction filtering (Method II, cf. above). The combined noise signal $es(n)$ is used as a second input to the algorithm part of the adaptive FBC-filter. Appropriate delays are inserted to 'align' the samples of the different signals. In the embodiment of FIG. 6a, the output signal $y(n)$ of the forward path gain unit $G(z,n)$ is connected to a masked noise generator (cf. FIG. 2a and the discussion above) comprising Noise shape and level unit (controlled by a Control unit) for estimating time-varying shaping filter $M(z,n)$, which filters white noise sequence $w(n)$ and provides as an output the masked noise signal $us(n)$, which is added to the output signal $y(n)$ of the forward path gain unit in SUM unit '+' to provide output signal $u(n)$, which is connected to the receiver. The masked noise signal $us(n)$ is delayed in delay unit Delay $DB(z)$ providing output $USD(n)$ which is connected to the $Fh$
filter estimation unit. The purpose of the delay of \( us(n) \) is to align the noise-signal samples of the two input signals \( usd(n) \) and \( es(n) \) to the \( Fh \) filter estimation unit for generating update filter coefficients to the variable filter part \( Fh(z,n) \) of the FBC-filter for estimating the feedback signal \( v(n) \). The other input \( es(n) \) of the \( Fh \) filter estimation unit is generated by an enhancement unit implementing a combination of noise retrieval based on long term prediction filtering (Method I) and binaural prediction filtering (Method II).

The processing of the signal on the input side in FIG. 6a is a combination of the two retrieval techniques considered separately above: long term prediction (LTP) filtering (cf. block Retrieval of feedback noise) and binaural prediction filtering (cf. block Binaural retrieval of feedback noise). The blocks Delay \( DE1(z) \), LE1 filter estimation and \( LE1(z,n) \) form the LTP filter considered above. The blocks have been described in section Noise retrieval based on long term prediction (method I above). The output of this filter, \( ex(n) \), consists ideally of signal components with a correlation time no longer than \( N_z \). The filter structure consisting of Delay \( DE2(z) \) and \( LE2(z,n) \) implements exactly the same filter as Delay \( DE1(z) \) and \( LE1(z,n) \). Specifically, \( DE2(z) = DE1(z) \), and \( LE2(z,n) \) is copied whenever \( LE1(z,n) \) is updated, so \( LE2(z,n)=LE1(z,n) \) at all times. Consequently, \( ycx(n) \) is the signal \( yc(n) \) received from the external sensor, filtered through the LTP filter. The signals \( ex(n) \) and \( ycx(n) \) now enter the binaural retrieval filter in a similar manner as \( e(n) \) and \( yc(n) \) did for the stand-alone binaural retrieval filter described in FIG. 5. As mentioned, \( ex(n) \) consists of "noise-like" components, some originating from the inserted noise (these are the components of interest in this context) and some intrinsically present in the input signal (these are interference components in the given context). The purpose of the binaural retrieval filter is to reject these interference components, such that, ideally, the signal \( es(n) \) contains the noise-like components originating from the introduced noise.

The outputs of the Retrieval of feedback noise block are a first signal \( ex(n) \) comprising the noise-like parts of the feedback corrected input signal \( e(n) \) and a second signal \( ycx(n) \) comprising the alternative microphone signal, which has been filtered in a copy of the LTP filter \( (DE1(z), LE1(z,n)) \) These
signals are connected to the Binaural retrieval of feedback noise block, the second signal $y_c x(n)$ to the algorithm and variable filter parts of the adaptive filter (LS filter estimation and $LB(z,n)$, respectively) and the first signal $ex(n)$ to delay unit Delay $DB(z)$. The output of the variable filter part $LB(z,n)$ is subtracted from the output of Delay $DB(z)$ in SUM unit '+' This output $es(n)$ of the Binaural retrieval of feedback noise block represents the combined retrieved noise and is connected to the (internal) LB filter estimation unit (and used in the estimate of the variable filter part $LB(z,n)$) as well as to the $Fh$ filter estimation unit and used for updating the variable filter part $Fh(z,n)$ of the adaptive feedback cancellation filter.

A Control unit is in one- or two-way communication with the Noise shape and level unit and the LB-, LE- and Fh- Filter Estimation units and the forward path gain unit $G(z,n)$.

The output signal $u(n)$ is connected to the variable filter part $Fh(z,n)$ of the adaptive FBC-filter. The electrical equivalent $F(z,n)$ of the leakage feedback from output to input transducer resulting in input signal $v(n)$ is added to a target signal $x(n)$ in SUM unit '+' representing the microphone. The feedback signal estimate $vh(n)$ resulting from the feedback estimation $Fh(z,n)$ is subtracted from the combined input $x(n) + v(n)$ in SUM unit '+' whose output, the feedback corrected input signal $e(n)$, is connected to the forward path gain unit $G(z,n)$ and to the Retrieval of feedback noise block (here specifically to the Delay $DE1(z)$ unit). The Retrieval of feedback noise block is in FIG. 6a represented by units enclosed by the dotted rectangle, i.e. including units implementing filter $D(z,n)$ and the update LE1 filter estimation unit as outlined above AND delay unit $DE2(z)$ and variable filter part $LE2(z,n)$ for delaying and filtering alternative microphone signal $y_c(n)$ before it enters the Binaural retrieval of feedback noise block.

Noise retrieval based on long term prediction filtering (Method I), on binaural prediction filtering (Method II), and on extraction of intrinsic noise-like signal components (Method C) combined with noise generation based on masked noise (Method A), and on perceptual noise substitution (Method B):
In the embodiment of a listening device shown in FIG. 6b, **processing on the output side** includes perceptual noise substitution performed on the output signal \( y(n) \) from the forward path gain unit \( G(z,n) \) by block PNS providing corresponding outputs \( ulp(n) \), \( ups(n) \), which in successive SUM units '//' (the first providing combined P/NS-output signal \( upx(n) = upl(n) + ups(n) \)) are combined with the masked noise signal \( ms(n) \) (Method A, cf. above) generated by block **Masked probe noise** to provide the output signal \( u(n) = upx(n) + ms(n) \). These noise generation methods are further combined with the extraction of intrinsic noise in block **Retrieval of intrinsic noise** (Method C, filter \( C(z,n) \), cf. above) from the output signal \( u(n) \) \( (\alpha = 0) \) OR from the unaltered signal parts \( ulp(n) \) from the PNS block \( (\alpha = 1) \) (OR from a combination of the two, cf. gain factor \( 0 < \alpha < 1 \)) to generate a resulting noise-like signal \( us(n) \), which is used as a first input to the algorithm part (\( Fh \) filter estimation) of the adaptive FBC-filter for estimating the feedback path. This is largely as shown in FIG. 2g and as described in connection therewith. In FIG. 6b **processing on the input side** includes that the noise in the feedback corrected input signal \( e(n) \) originating from the inserted noise on the output side is retrieved in enhancement unit **Retrieval of feedback noise** using long term prediction filtering (Method I, filter \( D(z,n) \), cf. above) and noise from an alternative microphone signal (e.g. from a contra lateral device, e.g. processed in processing unit \( Y \)) is retrieved in enhancement unit **Binaural retrieval of feedback noise** using binaural prediction filtering (Method II, cf. above). The resulting noise signal \( es(n) \) is used as a second input to the algorithm part of the adaptive FBC-filter. Appropriate delays are inserted to 'align' the samples of the different signals. This is largely as shown and described in connection with FIG. 6a above.

The output signal \( u(n) \) is connected to the variable filter part \( Fh(z,n) \) of the adaptive FBC-filter. The electrical equivalent \( F(z,n) \) of the leakage feedback from output to input transducer resulting in input signal \( v(n) \) is added to a target signal \( x(n) \) in SUM unit '//' representing the microphone. The feedback signal estimate \( vh(n) \) resulting from the feedback estimation \( Fh(z,n) \) is subtracted from the combined input \( x(n) + v(n) \) in SUM unit '//' whose output, the feedback corrected input signal \( e(n) \), is connected to the forward path gain unit \( G(z,n) \) and to the **Retrieval of feedback noise** block.
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.

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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CLAIMS

1. A listening device for processing an input sound to an output sound, the listening device comprising
   • an input transducer for converting an input sound to an electric input signal and defining an input side,
   • an output transducer for converting a processed electric output signal to an output sound and defining an output side,
   • a forward path being defined between the input transducer and the output transducer, and comprising a signal processing unit adapted for processing an SPU-input signal originating from the electric input signal and to provide a processed SPU-output signal, and
   • an electric feedback loop from the output side to the input side comprising
     o a feedback path estimation unit for estimating an acoustic feedback transfer function from the output transducer to the input transducer, and
     o an enhancement unit for extracting characteristics of an electric signal of the forward path and providing an estimated characteristics output;

   wherein the feedback path estimation unit is adapted to use the estimated characteristics output in the estimation of the acoustic feedback transfer function.

2. A listening device according to claim 1 wherein said feedback path estimation unit comprises an adaptive filter comprising a variable filter part and an algorithm part for updating filter coefficients of the variable filter part, the algorithm part being adapted to base the update at least partly on said estimated characteristics output from the enhancement unit.

3. A listening device according to claims 1 or 2 wherein the characteristics of the electric signal of the forward path are selected from the group comprising a modulation index, periodicity, correlation time, and combinations thereof.
4. A listening device according to any one of claims 1-3 wherein the input side of the forward path 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 of the input signal at consecutive points in time $t_n=n^*(1/f_s)$.

5. A listening device according to any one of claims 1-4 wherein the signal processing unit is adapted for processing the SPU-input signal originating from the electric input signal in frequency bands.

6. A listening device according to any one of claims 1-5, wherein the enhancement unit is adapted for retrieving intrinsic noise-like signal components in the electric signal of the forward path.

7. A listening device according to claim 6, wherein the correlation time $N_1$ of the noise signal estimate output obeys $N_1 < dG$, where $dG$ is the delay of the forward path.

8. A listening device according to claim 6 or 7 wherein the enhancement unit comprises an adaptive filter $C(z,n)$ of the form

$$C(z,n) = 1 - DR(z) \times LR(z,n)$$

$$z^{-1} \sum_{p=0}^{P_i} \sum_{p=1}^{N_1} c_p z^{-p},$$

where $C(z,n)$ represents the resulting filter, $DR(z) = z^{N1}$ represents a delay corresponding to $Ni$ samples, $LR(z,n)$ represents the variable filter part, $N_1$ is the maximum correlation time, and $c_p$ are the filter coefficients adapted to minimize a statistical correlation deviation measure of $us(n)$ and $us(n)$ is the noise signal estimate output, and where $P_i$ is the order of $LR(z,n)$.

9. A listening device according to any one of claims 1-8 comprising a probe signal generator for generating a probe signal contributing to the estimation of the feedback transfer function.
10. A listening device according to claim 9 wherein the probe signal generator is adapted to provide that the probe signal has predefined characteristics, and wherein the enhancement unit is adapted to provide a noise signal estimate output based on said characteristics.

11. A listening device according to claim 9 or 10 wherein the probe signal generator is adapted to provide that the probe signal has a correlation time $N_0$ which is smaller than or equal to the sum of the forward path and feedback path delays, e.g. $\leq 5$ ms, such as $\leq 64$ samples.

12. A listening device according to any one of claims 9-11 wherein the algorithm part of the feedback path estimation unit comprises a step length control block for controlling the step length of the algorithm in a given frequency region, and wherein the step length control block receives a control input from the probe signal generator.

13. A listening device according to any one of claims 9-12 wherein the probe signal generator is adapted to provide a probe signal based on masked added noise.

14. A listening device according to claim 13 wherein the probe signal generator comprises an adaptive filter for filtering a white noise input sequence $w$, the output of the variable part $M$ of the adaptive filter forming the masked probe signal, and the variable part $M$ of the adaptive filter being updated based on a signal from the forward path by an algorithm part comprising a model of the human auditory system.

15. A listening device according to any one of claims 9-14 wherein the probe signal generator is adapted to provide a probe signal based on perceptual noise substitution, PNS.

16. A listening device according to claim 15 wherein the probe signal generator comprises a PNS-part located in the forward path, and bases its output on a perceptual noise substitution algorithm for substituting one or more spectral regions of its input signal with filtered noise sequences.
17. A listening device according to any one of claims 9-16 wherein the enhancement unit is adapted to base the noise signal estimate output on an adaptive long-term prediction, LTP, filter \( D(z,n) \) adapted for filtering a feedback corrected input signal on the input side of the forward path to provide a noise signal estimate output comprising noise-like signal components said feedback corrected input signal.

18. A listening device according to claim 17 wherein the adaptive LTP filter \( D \) has a time varying filter characteristic and is of the specific form

\[
D(z,n) = 1 - DE(z) \times LE(z,n)
\]

\[
= 1 - z^{-N_2} \sum_{p=0}^{P_1} d_{p-N_2} z^{-p}
\]

\[
= 1 - \sum_{p=N_2}^{P_1} d_{p} z^{-p}
\]

where \( D(z,n) \) represents the resulting filter, \( DE(z) = z^{N_2} \) represents a delay corresponding to \( N_2 \) samples, \( LE(z,n) \) represents the variable filter part, \( N_2 \) is the maximum correlation time, \( d_p \) are the filter coefficients adapted to minimize a statistical deviation measure of \( es(n) \), and \( P_2 \) is the order of the filter \( LE(z,n) \), and where \( es(n) \) is the output signal of the filter \( D(z,n) \), and

\[
es(n) = e(n) - \sum_{l=0}^{P_2} d_{l} e(n - N2 - l) = e(n) - z(n),
\]

and \( e(n) \) is a feedback-corrected input signal on the input side at time instant \( n \).

19. A listening device according to claim 18 wherein \( N_2 \) is larger than 32, such as in the range between 40 and 200 samples for \( f_s = 20 \) kHz.

20. A listening device according to claim 18 wherein \( N_2 \) is larger than or equal to \( N_0 + N \), where \( N_0 \) represents the correlation time of the probe noise sequence, and \( N \) represents the efficient length in time of the feedback path impulse response.

21. A listening device according to any one of claims 9-20 wherein the enhancement unit is adapted to provide a noise signal estimate output.
based on binaural prediction filtering, wherein an adaptive noise retrieval filter E is adapted for filtering a signal $y_c$ from another microphone, e.g. from the input side of the forward path of a contra-lateral listening device.

22. A listening device according to claim 21 wherein the adaptive noise retrieval filter E has a time varying filter characteristic described by the difference equation

$$e_i(n) = e(n - N_3) - \sum_{p=0}^{P_3} e_p y_c(n - p),$$

where $y_c(n)$ represents samples from the other microphone, e.g. an external sensor, and

$$LB(z,n) = \sum_{p=0}^{P_3} e_p z^{-p}$$

represents the variable filter part, where $e_p$ are the filter coefficients adapted to minimize a statistical deviation measure of $es(n)$ and where, $N_3$ is a delay in samples and $P_3$ is the order of the filter $LB(z,n)$.

23. A listening device according to claim 22 wherein $N_3$ is in the range from 0 samples to 400 samples at a sampling frequency of 20 kHz.

24. A method of estimating a feedback transfer function in a listening device comprising a feedback estimation system for estimating acoustic feedback, the hearing device comprising

- a forward path between an input transducer and an output transducer and comprising a signal processing unit adapted for processing an SPU-input signal originating from the electric input signal and to provide a processed SPU-output signal $u$,

- an electric feedback loop from the output side to the input side comprising a feedback path estimation unit for estimating the feedback transfer function from the output transducer to the input transducer,

the method comprising

- extracting characteristics of the electric signal of the forward path and providing an estimated characteristics output;
• adapting the feedback path estimation unit to use the estimated characteristics output in the estimation of the feedback transfer function.

25. A method according to claim 24 wherein characteristics of the electric signal of the forward path comprises one or more of the following: modulation index, periodicity, correlation time, noise or noise-like parts.

26. A method according to claim 24 or 25 wherein extracting characteristics of the electric signal of the forward path comprises estimating signal components in the electric signal of the forward path originating from noise-like signal parts and the estimated characteristics output comprises a noise signal estimate output.

27. A method according to any one of claims 24-26 further comprising inserting noise-like signal parts in the forward path.

28. 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 according to any one of claims 24-27, when said computer program is executed on the data processing system.

29. 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 according to any one of claims 24-27.
FIG. 2b
FIG. 2d
FIG. 2e
A. CLASSIFICATION OF SUBJECT MATTER

INV. H04R25/00

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

H04R

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

EPO-Internal

C. DOCUMENTS CONSIDERED TO BE RELEVANT

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<th>Citation of document, with indication, where appropriate, of the relevant passages</th>
<th>Relevant to claim No</th>
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D. Further documents are listed in the continuation of Box C

See patent family annex

'*' Special categories of cited documents

'A' document defining the general state of the art which is not considered to be of particular relevance

'E' earlier document but published on or after the international filing date

'X' document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

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'P' document published prior to the international filing date but later than the priority date claimed

'T' later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

'X' document of particular relevance, the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

'Y' document of particular relevance, the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art

'S' document member of the same patent family

Date of the actual completion of the international search

29 March 2010

Date of mailing of the international search report

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