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(54) Noise reducing sound reproduction

(57) A noise reducing sound reproduction system and method is disclosed, in which: an input signal is supplied to a loudspeaker by which it is acoustically radiated; the signal radiated by the loudspeaker is received by a microphone that is acoustically coupled to the loudspeaker via a secondary path and that provides a microphone output signal; from the microphone output signal a useful-signal is subtracted to generate a filter input signal; the filter input signal is filtered in an active noise reduction filter to generate an error signal; and the useful-signal is subtracted from the error signal to generate the loudspeaker input signal; and the useful-signal is filtered by one or more spectrum shaping filters prior to subtraction from the microphone output signal or the loudspeaker input signal or both.

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Description

BACKGROUND

1. Field

[0001] Disclosed herein is a noise reducing sound reproduction system and, in particular, a noise reduction system which includes an earphone for allowing a user to enjoy, for example, reproduced music or the like, with reduced ambient noise.

2. Related Art

[0002] In active noise reduction systems, also known as active noise cancellation/control (ANC) systems, the same loudspeakers, in particular loudspeakers arranged in the two earphones of headphones, are often used for both noise reduction and reproduction of desirable sound such as music or speech. However, there is a significant difference between the sound impression created by employing active noise reduction and the impression created by not employing active noise reduction, due to the fact that common noise reduction systems reduce the desirable sound to a certain degree, as well. Accordingly, either advanced electrical signal processing is required to compensate for this effect or the listener has to accept sound impressions that differ, depending on whether noise reduction is on or off. Therefore, there is a general need for an improved noise reduction system to overcome this drawback.

SUMMARY OF THE INVENTION

[0003] In a first aspect of the invention, a noise reducing sound reproduction system is disclosed that comprises: a loudspeaker that is connected to a loudspeaker input path; a microphone that is acoustically coupled to the loudspeaker via a secondary path and connected to a microphone output path; a first subtractor that is connected downstream of the microphone output path and a first useful-signal path; an active noise reduction filter that is connected downstream of the first subtractor; and a second subtractor that is connected between the active noise reduction filter and the loudspeaker input path and to a second useful-signal path; in which both useful-signal paths are supplied with a useful signal to be reproduced and at least one of the useful-signal paths comprise one or more spectrum shaping filters.

[0004] In a second aspect of the invention, a noise reducing sound reproduction method is disclosed, in which: an input signal is supplied to a loudspeaker by which it is acoustically radiated; the signal radiated by the loudspeaker is received by a microphone that is acoustically coupled to the loudspeaker via a secondary path and that provides a microphone output signal; from the microphone output signal a useful-signal is subtracted to generate a filter input signal; the filter input signal is filtered in an active noise reduction filter to generate an error signal; and the useful-signal is subtracted from the error signal to generate the loudspeaker input signal; and the useful-signal is filtered by one or more spectrum shaping filters prior to subtraction from the microphone output signal or the loudspeaker input signal or both.

BRIEF DESCRIPTION OF THE DRAWINGS

[0005] Various specific embodiments are described in more detail below based on the exemplary embodiments shown in the figures of the drawing. Unless stated otherwise, similar or identical components are labeled in all of the figures with the same reference numbers.

FIG. 1 is a block diagram of a general feedback type active noise reduction system in which the useful signal is supplied to the loudspeaker signal path;

FIG. 2 is a block diagram of a general feedback type active noise reduction system in which the useful signal is supplied to the microphone signal path;

FIG. 3 is a block diagram of a general feedback type active noise reduction system in which the useful signal is supplied to the loudspeaker and microphone signal paths;

FIG. 4 is a block diagram of the active noise reduction system of FIG. 3, in which the useful signal is supplied via a spectrum shaping filter to the loudspeaker path.

FIG. 5 is a block diagram of the active noise reduction system of FIG. 3, in which the useful signal is supplied via a spectrum shaping filter to the microphone path;

FIG. 6 is a schematic diagram of an earphone applicable in connection with the active noise reduction systems of FIGS. 3-6;

FIG. 7 is a block diagram of the active noise reduction system of FIG. 5 in which the useful signal is supplied via two spectrum shaping filters to the microphone path;

FIG. 8 is a magnitude frequency response diagram representing the transfer characteristics of shelving filters applicable in the system of FIG. 7;

FIG. 9 is a magnitude frequency response diagram representing the transfer characteristics of equalizing filters applicable in the system of FIG. 7; and

FIG. 10 is a block diagram of the active noise reduction system of FIG. 5, in which the useful signal is supplied via spectrum shaping filters to the microphone and loudspeaker paths.

DETAILED DESCRIPTION

[0006] Feedback ANC systems are intended to reduce or even cancel a disturbing signal, such as noise, by providing at a listening site a noise reducing signal that ideally has the same amplitude over time but the opposite phase compared to the noise signal. By superimposing the noise signal and the noise reducing signal, the resulting signal, also known as error signal, ideally tends toward zero. The quality of the noise reduction depends on the quality of a so-called secondary path, i.e., the acoustic path between a loudspeaker and a microphone representing the listener's ear. The quality of the noise reduction further depends on the quality of a so-called ANC filter that is connected between the microphone and the loudspeaker and that filters the error signal provided by the microphone such that, when the filtered error signal is reproduced by the loudspeaker, it further reduces the error signal. However, problems occur when additionally to the filtered error signal a useful signal such as music or speech is provided at the listening site, in particular by the loudspeaker that also reproduces the filtered error signal. Then, the useful signal may be deteriorated by the system as previously mentioned.

[0007] For the sake of simplicity, no distinction is made herein between electrical and acoustic signals. However, all signals provided by the loudspeaker or received by the microphone are actually of an acoustic nature. All other signals are electrical in nature. The loudspeaker and the microphone may be part of an acoustic sub-system (e.g., a loudspeaker-room-microphone system) having an input stage formed by the loudspeaker 3 and an output stage formed by the microphone; the sub-system being supplied with an electrical input signal and providing an electrical output signal. "Path" means in this regard an electrical or acoustical connection that may include further elements such as signal conducting means, amplifiers, filters, etc. A spectrum shaping filter is a filter in which the spectra of the input and output signal are different over frequency.

[0008] Reference is now made to FIG. 1, which is a block diagram illustrating a general feedback type active noise reduction (ANC) system in which a disturbing signal d[n], also referred to as noise signal, is transferred (radiated) to a listening site, e.g., a listener's ear, via a primary path 1. The primary path 1 has a transfer characteristic of P(z). Additionally, an input signal v[n] is transferred (radiated) from a loudspeaker 3 to the listening site via a secondary path 2. The secondary path 2 has a transfer characteristic of S(z).

[0009] A microphone 4 positioned at the listening site receives together with the disturbing signal d[n] the signals that arise from the loudspeaker 3. The microphone 4 provides a microphone output signal y[n] that represents the sum of these received signals. The microphone output signal y[n] is supplied as filter input signal u[n] to an ANC filter 5 that outputs to an adder 6 an error signal e[n]. The ANC filter 5, which may be an adaptive filter, has a transfer characteristic of W(z). The adder 6 also receives an optionally pre-filtered, e.g., with a spectrum shaping filter (not shown in the drawings) useful signal x[n] such as music or speech and provides an input signal v[n] to the loudspeaker 3.

[0010] The signals x[n], y[n], e[n], u[n] and v[n] are in the discrete time domain. For the following considerations their spectral representations X(z), Y(z), E(z), U(z) and V(z) are used. The differential equations describing the system illustrated in FIG. 1 are as follows:

$$Y(z) = S(z) \cdot V(z) = S(z) \cdot (E(z)+X(z)) \tag{1}$$

$$E(z) = W(z) \cdot U(z) = W(z) \cdot Y(z) \tag{2}$$

[0011] In the system of FIG. 1, the useful signal transfer characteristic M(z) = Y(z)/X(z) is thus

$$M(z) = S(z)/(1-W(z)\cdot S(z)) \qquad (3)$$

[0012] Assuming W(z) = 1 then

$$\lim[S(z)\rightarrow 1]\ M(z) \Rightarrow M(z)\rightarrow\infty \qquad (4)$$

$$\lim[S(z)\rightarrow\pm\infty]\ M(z) \Rightarrow M(z)\rightarrow 1 \qquad (5)$$

$$\lim[S(z)\rightarrow 0]\ M(z) \Rightarrow S(z \qquad (6)$$

[0013] Assuming W(z) = ∞ then

$$\lim[S(z)\rightarrow 1]\ M(z) \Rightarrow M(z)\rightarrow 0. \qquad (7)$$

[0014] As can be seen from equations (4)-(7), the useful signal transfer characteristic M(z) approaches 0 when the transfer characteristic W(z) of the ANC filter 5 increases, while the secondary path transfer function S(z) remains neutral, i.e. at levels around 1, i.e., 0[dB]. For this reason, the useful signal x[n] has to be adapted accordingly to ensure that the useful signal x[n] is apprehended identically by a listener when ANC is on or off. Furthermore, the useful signal transfer characteristic M(z) also depends on the transfer characteristic S(z) of the secondary path 2 to the effect that the adaption of the useful signal x[n] also depends on the transfer characteristic S(z) and its fluctuations due to aging, temperature, change of listener etc. so that a certain difference between "on" and "off" will be apparent.

[0015] While in the system of FIG. 1 the useful signal x[n] is supplied to the acoustic sub-system (loudspeaker, room, microphone) at the adder 6 connected upstream of the loudspeaker 3, in the system of FIG. 2 the useful signal x[n] is supplied at the microphone 4. Therefore, in the system of FIG. 2, the adder 6 is omitted and an adder 7 is arranged downstream of microphone 4 to sum up the, e.g., pre-filtered, useful signal x[n] and the microphone output signal y[n]. Accordingly, the loudspeaker input signal v[n] is the error signal [e], i.e., v[n] = [e], and the filter input signal u[n] is the sum of the useful signal x[n] and the microphone output signal y[n], i.e., u[n] = x[n]+y[n].

[0016] The differential equations describing the system illustrated in FIG. 2 are as follows:

$$Y(z) = S(z)\cdot V(z) = S(z)\cdot E(z) \qquad (8)$$

$$E(z) = W(z)\cdot U(z) = W(z)\cdot(X(z)+Y(z)) \qquad (9)$$

[0017] The useful signal transfer characteristic M(z) in the system of FIG. 2 without considering the disturbing signal d[n] is thus

$$M(z) = (W(z)\cdot S(z))/(1-W(z)\cdot S(z)) \qquad (10)$$

$$\lim[(W(z)\cdot S(z))\to 1]\ M(z)\ \Rightarrow M(z)\to\infty \tag{11}$$

$$\lim[(W(z)\cdot S(z))\to 0]\ M(z)\ \Rightarrow M(z)\to 0 \tag{12}$$

$$\lim[(W(z)\cdot S(z))\to\pm\infty]\ M(z)\ \Rightarrow M(z)\to 1. \tag{13}$$

[0018] As can be seen from equations (11)-(13), the useful signal transfer characteristic M(z) approaches 1 when the open loop transfer characteristic (W(z) · S(z)) increases or decreases and approaches 0 when the open loop transfer characteristic (W(z) · S(z)) approaches 0. For this reason, the useful signal x[n] has to be adapted additionally in higher spectral ranges to ensure that the useful signal x[n] is apprehended identically by a listener when ANC is on or off. Compensation in higher spectral ranges is, however, quite difficult so that a certain difference between "on" and "off" will be apparent. On the other hand, the useful signal transfer characteristic M(z) does not depend on the transfer characteristic S(z) of the secondary path 2 and its fluctuations due to aging, temperature, change of listener etc.

[0019] FIG. 3 is a block diagram illustrating a general feedback type active noise reduction system in which the useful signal is supplied to both, the loudspeaker path and the microphone path. For the sake of simplicity, the primary path 1 is omitted below notwithstanding that noise (disturbing signal d[n]) is still present. In particular, the system of FIG. 3 is based on the system of FIG. 1, however, with an additional subtractor 8 that subtracts the useful signal x[n] from the microphone output signal y[n] to form the ANC filter input signal u[n] and with a subtractor 9 that substitutes adder 6 and subtracts the useful signal x[n] from error signal e[n].

[0020] The differential equations describing the system illustrated in FIG. 3 are as follows:

$$Y(z) = S(z)\cdot V(z) = S(z)\cdot(E(z)-X(z)) \tag{14}$$

$$E(z) = W(z)\cdot U(z) = W(z)\cdot(Y(z)-X(z)) \tag{15}$$

[0021] The useful signal transfer characteristic M(z) in the system of FIG. 3 is thus

$$M(z) = (S(z)-W(z)\cdot S(z))/(1-W(z)\cdot S(z)) \tag{16}$$

$$\lim[(W(z)\cdot S(z))\to 1]\ M(z)\ \Rightarrow M(z)\to\infty \tag{17}$$

$$\lim[(W(z)\cdot S(z))\to 0]\ M(z)\ \Rightarrow M(z)\to S(z) \tag{18}$$

$$\lim[(W(z)\cdot S(z))\to\pm\infty]\ M(z)\ \Rightarrow M(z)\to 1. \tag{19}$$

[0022] It can be seen from equations (17)-(19) that the behavior of the system of FIG. 3 is similar to that of the system of FIG. 2. The only difference is that the useful signal transfer characteristic M(z) approaches S(z) when the open loop transfer characteristic (W(z) · S(z)) approaches 0. Like the system of FIG. 1, the system of FIG. 3 depends on the transfer characteristic S(z) of the secondary path 2 and its fluctuations due to aging, temperature, change of listener etc.

[0023] In FIG. 4, a system is shown that is based on the system of FIG. 3 and that additionally includes an equalizing filter 10 connected upstream of the subtractor 9 in order to filter the useful signal x[n] with the inverse secondary path transfer function 1/S(z). The differential equations describing the system illustrated in FIG. 4 are as follows:

$$Y(z) = S(z) \cdot V(z) = S(z) \cdot (E(z) - X(z)/S(z)) \qquad (20)$$

$$E(z) = W(z) \cdot U(z) = W(z) \cdot (Y(z) - X(z)) \qquad (21)$$

[0024] The useful signal transfer characteristic M(z) in the system of FIG. 4 is thus

$$M(z) = (1 - W(z) \cdot S(z)) / (1 - W(z) \cdot S(z)) = 1 \qquad (22)$$

[0025] As can be seen from equation (22), the microphone output signal y[n] is identical to the useful signal x[n], which means that signal x[n] is not altered by the system if the equalizer filter is exact the inverse of the secondary path transfer characteristic S(z). The equalizer filter 10 may be a minimum-phase filter for optimum results, i.e., optimum approximation of its actual transfer characteristic to the inverse of, the ideally minimum phase, secondary path transfer characteristic S(z) and, thus y[n] = x[n]. This configuration acts as an ideal linearizer, i.e. it compensates for any deteriorations of the useful signal due to its transfer from the loudspeaker 3 to the microphone 4 representing the listener's ear. It hence compensates for, or linearizes the disturbing influence of the secondary path S(z) to the useful signal x[n], such that the useful signal arrives at the listener as provided by the source, without any negative effect due to acoustical properties of the headphone, i.e., y[z] = x[z]. As such, with the help of such a linearizing filter it is possible to make a poorly designed headphone sound like an acoustically perfectly adjusted, i.e. linear one.

[0026] In FIG. 5, a system is shown that is based on the system of FIG. 3 and that additionally includes an equalizing filter 10 connected upstream of the subtractor 8 in order to filter the useful signal x[n] with the secondary path transfer function S(z).

[0027] The differential equations describing the system illustrated in FIG. 5 are as follows:

$$Y(z) = S(z) \cdot V(z) = S(z) \cdot (E(z) - X(z)) \qquad (23)$$

$$E(z) = W(z) \cdot U(z) = W(z) \cdot (Y(z) - S(z) \cdot X(z)) \qquad (24)$$

[0028] The useful signal transfer characteristic M(z) in the system of FIG. 5 is thus

$$M(z) = S(z) \cdot (1 + W(z) \cdot S(z)) / (1 + W(z) \cdot S(z)) = S(z) \qquad (25)$$

[0029] From equation (25) it can be seen that the useful signal transfer characteristic M(z) is identical with the secondary path transfer characteristic S(Z) when the ANC system is active. When the ANC system is not active, the useful signal transfer characteristic M(z) is also identical with the secondary path transfer characteristic S(Z). Thus, the aural impression of the useful signal for a listener at a location close to the microphone 4 is the same regardless of whether noise reduction is active or not.

[0030] The ANC filter 5 and the equalizing filters 10 and 11 may be fixed filters with constant transfer characteristics or adaptive filters with controllable transfer characteristics. In the drawings, the adaptive structure of a filter per se is indicated by an arrow underlying the respective block and the optionality of the adaptive structure is indicated by a broken line.

[0031] The system shown in FIG. 5 is, for example, applicable in headphones in which useful signals, such as music or speech, are reproduced under different conditions in terms of noise and the listener may appreciate being able to switch off the ANC system, in particular when no noise is present, without experiencing any audible difference between the active and non-active state of the ANC system. However, the systems presented herein are not applicable in headphones only, but also in all other fields in which occasional noise reduction is desired.

[0032] FIG. 6 illustrates an exemplary earphone with which the present active noise reduction systems may be used. The earphone may be, together with another identical earphone, part of a headphone (not shown) and may be acoustically coupled to a listener's ear 12. In the present example, the ear 12 is exposed via primary path 1 to the disturbing signal d[n], e.g., ambient noise. The earphone comprises a cup-like housing 14 with an aperture 15 that may be covered by a sound permeable cover, e.g., a grill, a grid or any other sound permeable structure or material. The loudspeaker 3 radiates sound to the ear 12 and is arranged at the aperture 15 of the housing 14, both forming an earphone cavity 13. The cavity 13 may be airtight or vented by any means, e.g., by means of a port, vent, opening, etc. The microphone 4 is positioned in front of the loudspeaker 3. An acoustic path 17 extends from the speaker 3 to the ear 12 and has a transfer characteristic which is approximated for noise control purposes by the transfer characteristic of the secondary path 2 which extends from the loudspeaker 3 to the microphone 4.

[0033] In mobile devices such as headphones, the space and energy available for the ANC system is quite limited. Digital circuitry may be too space and energy consuming and in mobile devices analog circuitry is often the preferred in the design of ANC systems. However, analog circuitry allows only for a very limited complexity of the ANC system and thus it is hard to correctly model the secondary path solely by analog means. In particular, analog filters used in an ANC system are often fixed filters or very simple adaptive filters because they are easy to build, have low energy consumption and require little space. The systems illustrated above with reference to FIGS. 4, 5 and 7 also provide good results when employing analog circuitry as there is a minor (FIG. 4) or even no (FIGS. 5 and 7) dependency on the secondary path behavior. Furthermore, the systems of FIGS. 5 and 7 allow for a good estimation of the necessary transfer characteristic of the equalization filter based on the ANC filter transfer characteristic $W(z)$, as well as on the secondary path filter characteristic $S(z)$, both forming the open loop transfer characteristic $W(z) \cdot S(z)$, which, in principal, has only minor fluctuations, and based on the assessment of the acoustic properties of the headphone when attached to a listener's head.

[0034] The ANC filter 5 will usually have a transfer characteristic that tends to have lower gain at lower frequencies with an increasing gain over frequency to a maximum gain followed by a decrease of gain over frequency down to loop gain. With high gain of the ANC filter 5, the loop inherent in the ANC system keeps the system linear in a frequency range of, e.g., below 1 kHz and thus renders any equalization redundant in this frequency range. In the frequency range above 3 kHz, the ANC filter 5 has almost no boosting or cutting effects and, accordingly, no linearization effects. As the ANC filter gain in this frequency range is approximately loop gain, the useful signal transfer characteristic $M(z)$ experiences a boost at higher frequencies that has to be compensated for by means of a respective filter, e.g. a shelving filter, additional to an equalizing filter. In the frequency range between 1 kHz and 3 kHz both, boosts and cuts, may occur. In terms of aural impression, boosts are more disturbing than cuts and thus it may be sufficient to compensate for boosts in the transfer characteristic by correspondingly designed cut filters. If the ANC filter gain is 0 dB above 3 kHz, there is no linearization effect and, therefore, in addition to a first equalization filter and instead a shelving filter, a second equalizing filter may be used.

[0035] As can be seen from the above considerations, at least two filters may be used for compensation. FIG. 7 shows an exemplary ANC system that employs (at least) two filters 18 and 19 (sub-filters) instead of a single filter 11 as in the system of FIG. 5. For instance, a treble cut shelving filter (e.g., filter 18) having a transfer characteristic $S_1(z)$ and a treble cut equalizing filter (e.g., filter 19) having a transfer characteristic $S_2(z)$, in which $S(z) = S_1(z) \cdot S_2(z)$. Alternatively, a treble boost equalizing filter may be implemented as, e.g., filter 18 and a treble cut equalizing filter as, e.g., filter 19. If the useful signal transfer characteristic $M(z)$ exhibits an even more complex structure, three filters may be employed, e.g., one treble cut shelving filter and two treble boost/cut equalizing filters. The number of filters used may depend on many other aspects such as costs, noise behavior of the filters, acoustic properties of the headphone, delay time of the system, space available for implementing the system, etc.

[0036] FIG. 8 is a schematic diagram of the transfer characteristics a, b of shelving filters applicable in the system of FIG. 7. In particular, a first order treble boost (+9 dB) shelving filter (a) and a bass cut (-3 dB) shelving filter (b) are shown. FIG. 9 is a schematic diagram of the transfer characteristics c, d of equalizing filters applicable in the system of FIG. 7. One (c) of the equalizing filters provides a 9 dB boost at 1 kHz and the other (d) a 6 dB cut at 100 Hz having a higher Q and, thus, a sharper bandwidth.

[0037] Although the range of spectrum shaping functions is governed by the theory of linear filters, the adjustment of those functions and the flexibility with which they can be adjusted varies according to the topology of the circuitry and the requirements that have to be fulfilled. Shelving filters are usually simple first-order filters which alter the relative gains between frequencies much higher and much lower than the corner frequencies. A low or bass shelf is adjusted to affect the gain of lower frequencies while having no effect well above its corner frequency. A high or treble shelf adjusts the gain of higher frequencies only. A single equalizer filter, on the other hand, implements a second-order filter function.

This involves three adjustments: selection of the center frequency, adjustment of the quality (Q) factor, which determines the sharpness of the bandwidth, and the level or gain which determines how much the selected center frequency is boosted or cut relative to frequencies (much) above or below the center frequency.

[0038] FIG. 10 is a combination of the systems shown in FIGS. 4 and 5 in which the useful signal x[n] is supplied to both, the microphone and the loudspeaker path, each via a filter 20 having a transfer characteristic $S_5(z)$ or a filter 21 having a transfer characteristic $S_6(z)$, in which, for instance, $S(z) = S_5(z) \cdot S_6(z)$.

[0039] Although various examples of realizing the invention have been disclosed, it will be apparent to those skilled in the art that various changes and modifications can be made which will achieve some of the advantages of the invention without departing from the spirit and scope of the invention. It will be obvious to those reasonably skilled in the art that other components performing the same functions may be suitably substituted. Such modifications to the inventive concept are intended to be covered by the appended claims.

Claims

1. A noise reducing sound reproduction system comprising:

a loudspeaker that is connected to a loudspeaker input path;
a microphone that is acoustically coupled to the loudspeaker via a secondary path and connected to a microphone output path;
a first subtractor that is connected downstream of the microphone output path and a first useful-signal path;
an active noise reduction filter that is connected downstream of the first subtractor; and
a second subtractor that is connected between the active noise reduction filter and the loudspeaker input path and to a second useful-signal path; in which
both useful-signal paths are supplied with a useful signal to be reproduced and
at least one of the useful-signal paths comprises one or more spectrum shaping filters.

2. The system of claim 1, in which the secondary path has a secondary path transfer characteristic and at least one of the spectrum shaping filters has a transfer characteristic that models the secondary path transfer characteristic or linearizes a microphone signal on the microphone output path with regard to the useful signal.

3. The system of claim 1 or 2, in which the first useful-signal path comprises a first spectrum shaping filter that has a transfer characteristic that is identical with the secondary path transfer characteristic.

4. The system of claim 3, in which the first spectrum shaping filter comprises at least two sub-filters.

5. The system of claim 3 or 4, in which the first filter or at least one of the sub-filters of the first filter is an equalizing filter.

6. The system of one of claims 3-5, in which the at least one equalizing filter is a treble cut equalizing filter.

7. The system of claim 4 or 5, in which the first filter or one of the sub-filters of the first filter is a shelving filter.

8. The system of claim 7, in which the shelving filter is a treble cut shelving filter.

9. The system of claim 2, in which the second useful-signal path comprises a second spectrum shaping filter that has a transfer characteristic that is identical with the inverse secondary path transfer characteristic.

10. The system of one of claims 1-9, in which at least one of the active noise reduction filters, first spectrum shaping filter, and second shaping filter is an adaptive filter.

11. A noise reducing sound reproduction method, in which:

an input signal is supplied to a loudspeaker by which it is acoustically radiated;
the signal radiated by the loudspeaker is received by a microphone that is acoustically coupled to the loudspeaker via a secondary path and that provides a microphone output signal;
the microphone output signal is subtracted from a useful-signal to generate a filter input signal;
the filter input signal is filtered in an active noise reduction filter to generate an error signal; and
the useful-signal is subtracted from the error signal to generate the loudspeaker input signal; and

the useful-signal is filtered by one or more spectrum shaping filters prior to subtraction from the microphone output signal or the loudspeaker input signal or both.

12. The method of claim 11, in which the secondary path has a secondary path transfer characteristic and all spectrum shaping filters model in total the secondary path transfer characteristic.

13. The method of claim 12, in which the useful-signal, prior to subtraction from the microphone output signal, is filtered with a transfer characteristic that is identical with the secondary path transfer characteristic.

14. The method of claim 13, in which the filtering of the useful signal includes equalizing and/or shelving filtering.

15. The method of claim 12, in which the useful-signal, prior to subtraction from the loudspeaker input signal, is filtered with a transfer characteristic that is identical with the inverse secondary path transfer characteristic.

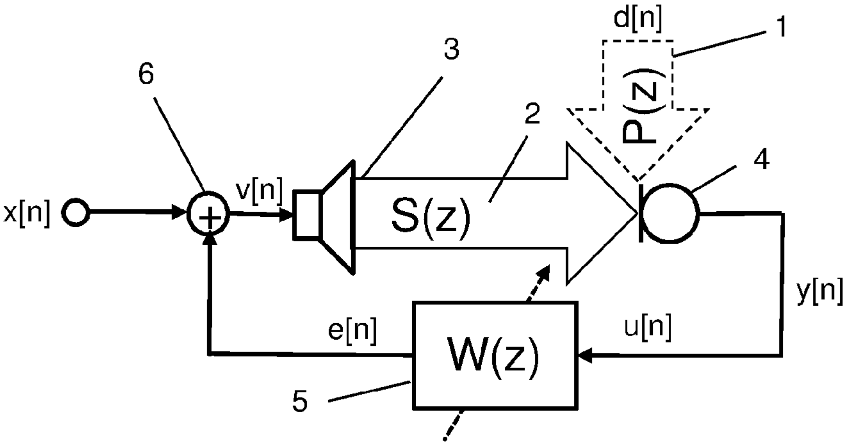


FIG 1

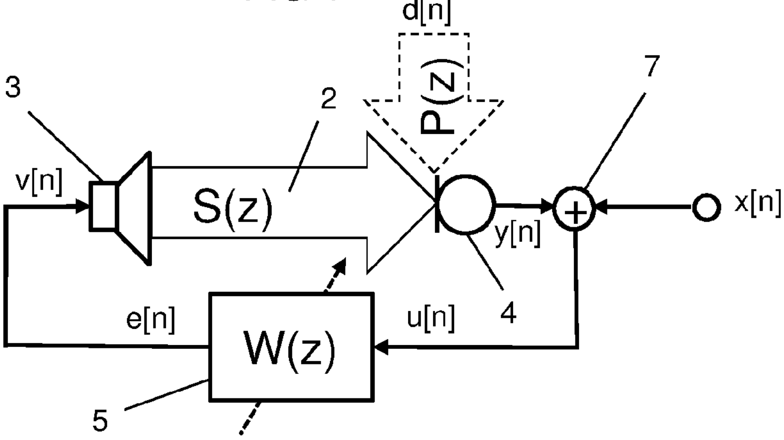


FIG 2

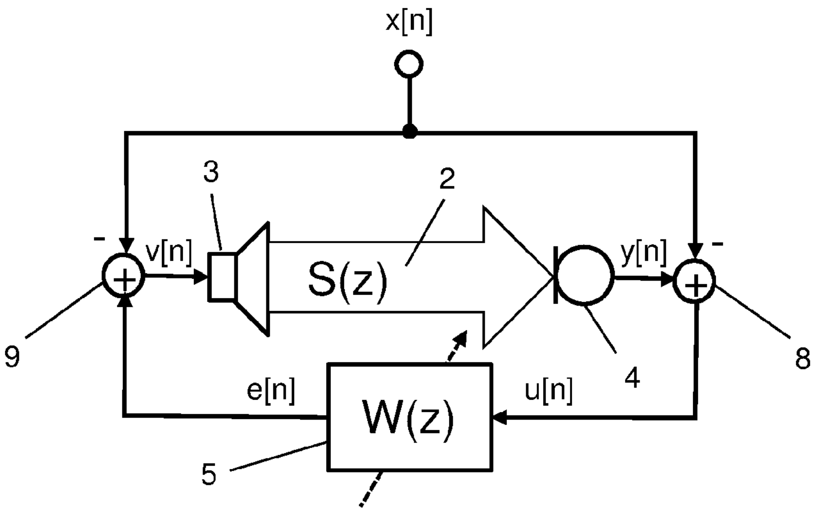


FIG 3

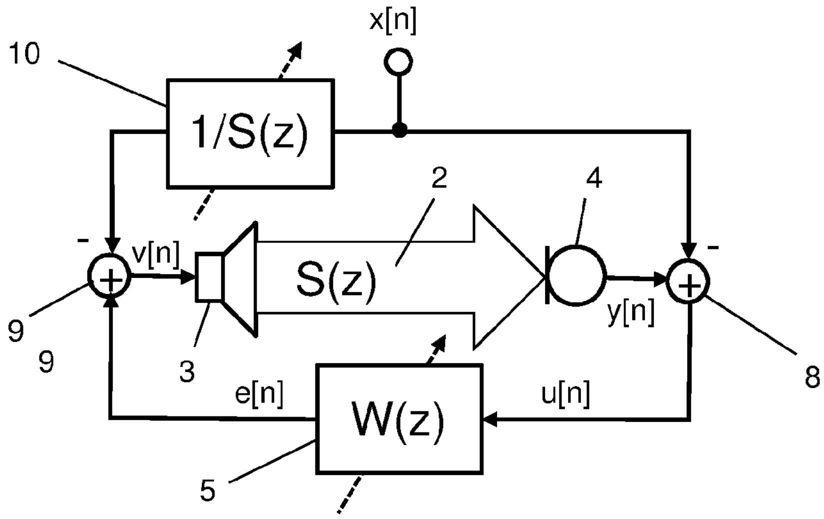
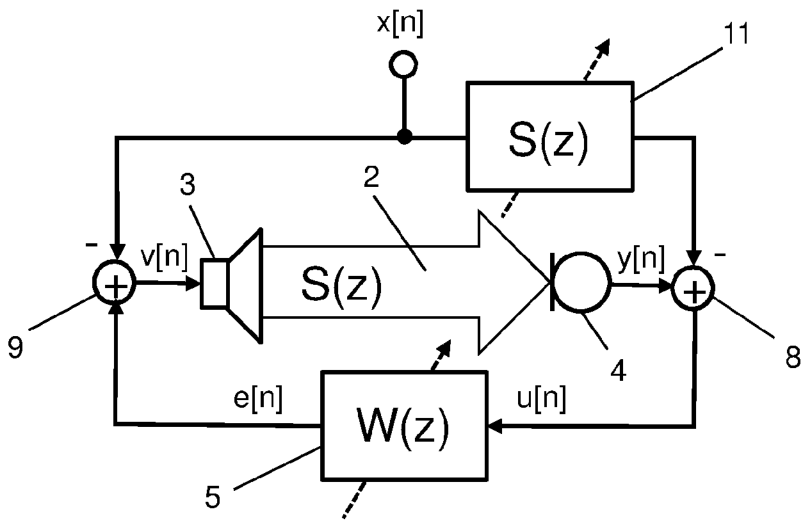

x[n]
10
1/S(z)
2
4
-
v[n]
S(z)
-
9
y[n]
9
3
8
e[n]
W(z)
u[n]
5

FIG 4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11
S(z)
3
2
-
v[n]
y[n]
-
S(z)
9
4
8
e[n]
u[n]
W(z)
5

FIG 5

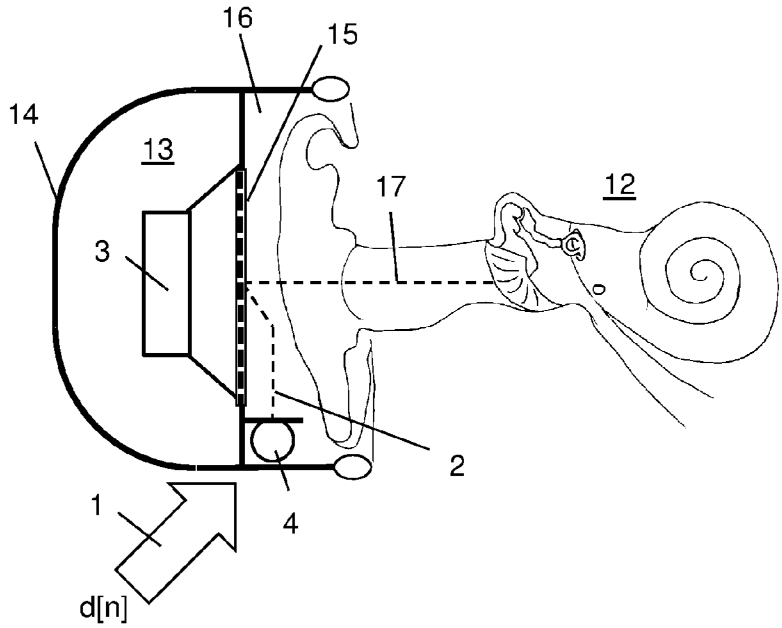


FIG 6

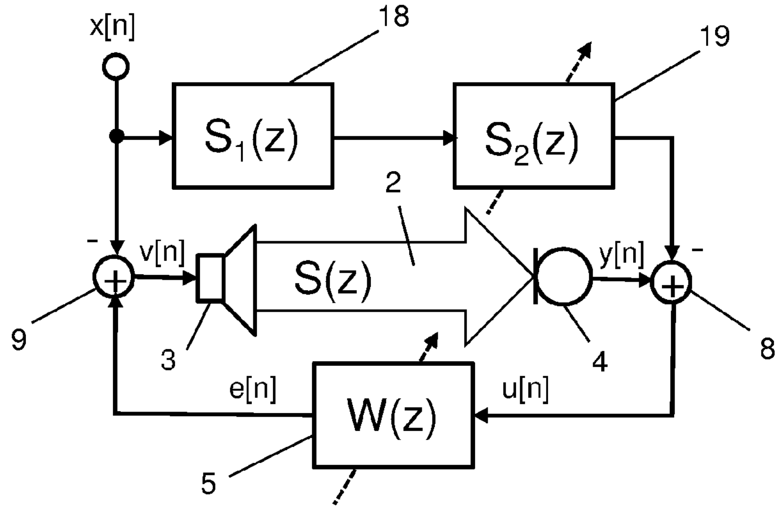


FIG 7

FIG 8

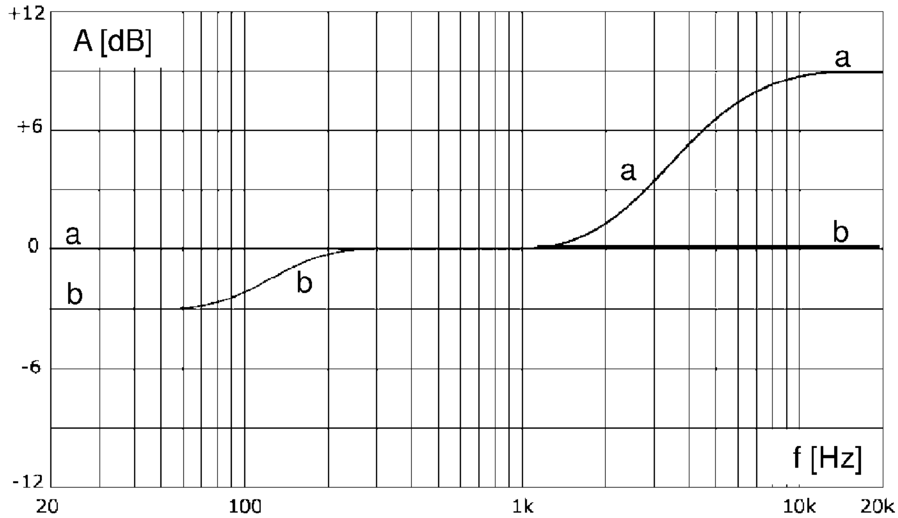


FIG 9

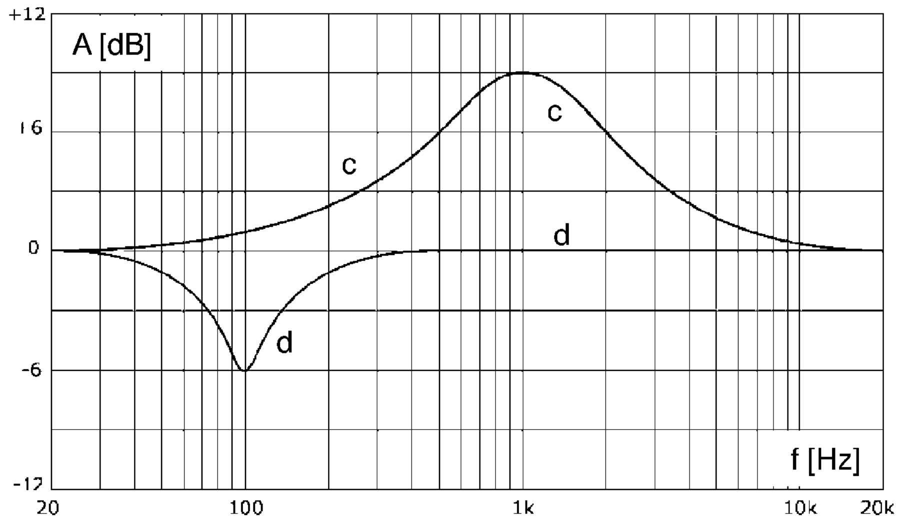
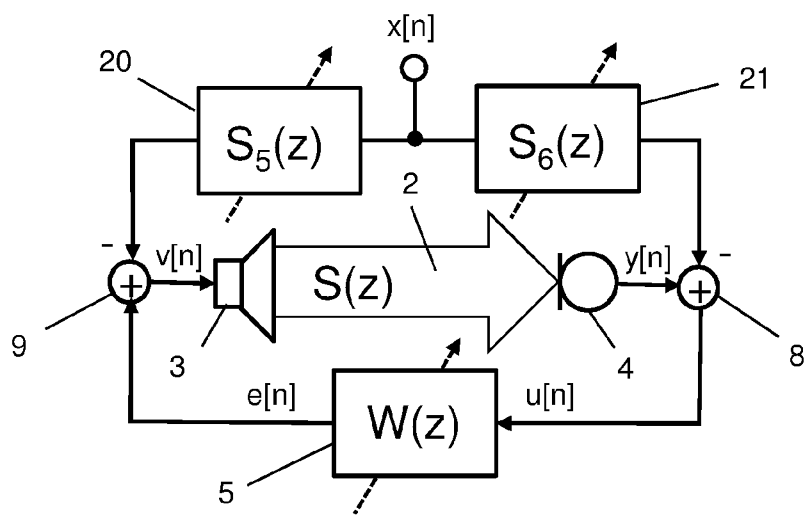


FIG 10



EUROPEAN SEARCH REPORT

Application Number
EP 11 17 5344

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A	US 2010/329473 A1 (RAUHALA JUKKA VESA TAPANI [FI]) 30 December 2010 (2010-12-30) * page 2, paragraph 0036 * -----	7,8,13	
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The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
The Hague	24 February 2012	Gastaldi, Giuseppe

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