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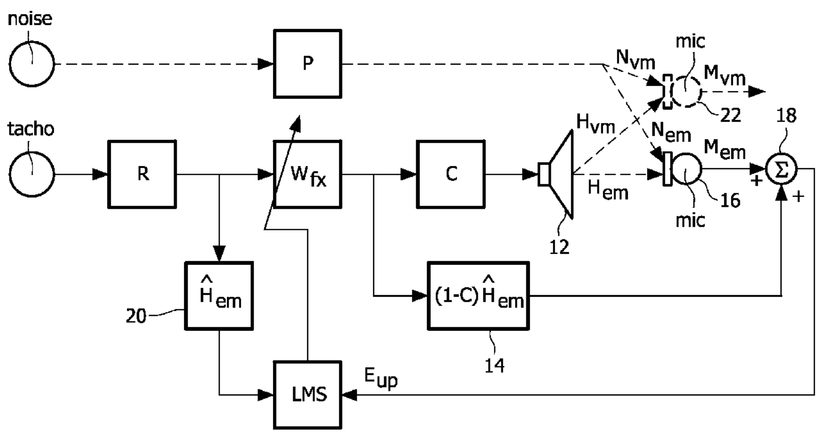
(54) **Title:** ACTIVE NOISE REDUCTION SYSTEM AND METHOD USING A VIRTUAL MICROPHONE



(57) **Abstract:** A signal processing system for active sound control comprising a reference signal generator (R), an adaptive filter (W_{fx}) arranged to receive an output of the reference signal generator (R), a loudspeaker (12) arranged to receive an output of the adaptive filter (W_{fx}), a first filter (14), also arranged to receive an output of the adaptive filter (W_{fx}), a microphone (16), a summer (18) arranged to receive outputs of the microphone (16) and the first filter (14), a second filter (20) arranged to receive an output of the reference signal generator (R), and a filter control (LMS) arranged to receive outputs of the second filter (20) and the summer (18), the filter control (LMS) arranged to control the adaptive filter (W_{fx}). The first filter (14) is arranged to filter the output of the adaptive filter (W_{fx}) according to a function of the ratio $C=A_{wf}/B_{vm}$, where A_{wf} is the ratio between a primary noise field at the microphone (16) and a primary noise field at a virtual microphone (22), and B_{vm} is the ratio between an acoustic path from the loudspeaker (12) to the microphone (16) and an acoustic path from the loudspeaker (12) to a virtual microphone (22).

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ACTIVE NOISE REDUCTION SYSTEM AND METHOD USING A VIRTUAL MICROPHONE

This invention relates to a signal processing system and to a method of operating the signal processing system. The signal processing system is particularly suitable for use in an active noise cancellation system, for example in a vehicle.

When traveling in a vehicle such as a car, the noise from the engine and from the road can be a discomfort to the driver and to the passengers in the vehicle. The noise that is present can also have a safety implication, if the driver is distracted while they are driving or fails to hear the approach of another vehicle or the presence of a warning signal from another vehicle. One possible solution to this problem is the use of active noise cancellation.

The principles of active noise cancellation (ANC) are already well known for a fairly long time. The theory, algorithms and applications are extensively treated in, for example "Active Noise Control Systems", by S. M. Kuo and D. R. Morgan, published by John Wiley, 1996. As an example, an algorithm is described in Chapter 4 of this book, which provides narrow band feed-forward active noise control. The goal of the algorithm is to have local suppression of engine noise (tonal noise) at the ears of the vehicle driver using a multi-frequency ANC FX-LMS algorithm.

Figure 1 illustrates this well known feed-forward ANC structure using the FX-LMS. The engine noise (primary wave field) produced during operation travels through the car (modeled by the transfer function P) and arrives at the driver's ears. Close to the driver's ears, the error microphone (M_{em}) picks up this noise (besides the actively produced secondary wave field) and the detected signal is used to adapt the filters of the FX-LMS algorithm. In addition to the signal from the error microphone, the LMS update rule also receives a filtered version of the reference signal.

In the example of Figure 1, the reference signal is a multi-tonal reference, synthesized from a tacho signal that shows a high coherence with the engine noise. This reference signal is filtered by an estimate of the secondary path before entering the LMS updating part. The secondary path (H_{em}) is the acoustical path from the secondary noise source (the loudspeaker) to the error microphone. For a correct working of the Active Noise

Canceller, the estimation of the secondary path is critical. Usually, this estimation is done in advance (off-line) and during operation the secondary path estimation is allowed to slowly adapt along changes in the environment.

With this algorithm, it is indeed possible to reduce the noise level locally around the error-microphone. A problem with this solution is that the error microphone is usually located a certain distance away (10-15 cm) from the driver's ears. The result of is that the ears of the driver are not in the sweet spot (where the most noise reduction is taking place) around the error-microphone. This problem has already been recognized in the past and solutions have also been suggested.

For example, United States Patent US 5381485 discloses an active sound control system comprising a loudspeaker having an input and operable to generate sound waves for interference with unwanted sound so as to produce a region close to the user of the system in which the perceived sound is substantially reduced. A monitoring microphone is positioned closer to the loudspeaker than to the region of sound reduction. Loudspeaker control means for controlling the input to the loudspeaker operate to energize the loudspeaker such that the sound waves emitted by the loudspeaker substantially cancel the unwanted sound waves in said region. The loudspeaker control means includes a signal processing means arranged to simulate a microphone output that would be obtained if that microphone, instead of being positioned closer to the loudspeaker than the user, were to be positioned in a notional position relatively close to the user. The resulting simulated or virtual microphone output is then used to control the signal fed to the loudspeaker input.

The concept of a virtual microphone has been introduced to solve the sweet spot problem. Moving the error microphone closer to the driver's head has been considered as not an acceptable solution. With the principle of a virtual microphone, it is possible to shift the sweet spot of the error-microphone into the direction of the driver's ears. The algorithm still adapts using the error-microphone, but with an altered transfer function.

An interpretation of the virtual microphone algorithm is given in Figure 2, which shows a feed-forward ANC structure using the FX-LMS with the virtual microphone. Next to the error microphone, there is also (temporarily) a microphone that is placed at (or very close to) the likely positions of the driver's ears. In advance, the acoustic path (H_{vm}) from the secondary loudspeaker to this virtual microphone will be estimated. During actual operation of this noise cancellation system, this microphone is not present any more, but the estimated path (H_{vm}) will be used to modify the update part of the FX-LMS algorithm. In Figure 2 this is represented by an extra signal flow branch or bridge. The result is that the sweet spot will move away from the error microphone into the direction of the driver's head, giving a better ANC effect at the ears.

There are, however, weaknesses with the ANC system shown in Figure 2. The virtual microphone algorithm as invented by Elliott and repeated in Figure 2 is based on an assumption that is not always true. The assumption is that, if the wavelength of the sound is large compared to the microphone separation distance (M_{em} and M_{vm}), the primary sound pressure on both microphones is equal. In many practical sound fields, like in the car, this assumption is not true. Certainly not for higher frequencies, which are still in a sensible range for ANC.

It is therefore an object of the invention to improve upon the known art.

According to a first aspect of the present invention, there is provided a signal processing system for active sound control comprising a reference signal generator, an adaptive filter arranged to receive an output of the reference signal generator, a loudspeaker arranged to receive an output of the adaptive filter, a first filter, also arranged to receive an output of the adaptive filter, a microphone, a summer arranged to receive outputs of the microphone and the first filter, a second filter arranged to receive an output of the reference signal generator, and a filter control arranged to receive outputs of the second filter and the summer, the filter control arranged to control the adaptive filter, wherein the first filter is arranged to filter the output of the adaptive filter according to a function of the ratio $C=A_{wf}/B_{vm}$, where A_{wf} is the ratio between a primary noise field at the microphone and a primary noise field at a virtual microphone, and B_{vm} is the ratio between an acoustic path from the loudspeaker to the microphone and an acoustic path from the loudspeaker to a virtual microphone.

According to a second aspect of the present invention, there is provided a method of operating a signal processing system for active sound control comprising; generating a reference signal, filtering, at an adaptive filter, the reference signal, outputting, at a loudspeaker, the output of the adaptive filter, filtering, at a first filter, the output of the adaptive filter, receiving, at a microphone, the output of the loudspeaker, combining, at a summer, the outputs of the microphone and the first filter, filtering, at a second filter, the reference signal, receiving, at a filter control, the outputs of the second filter and the summer, and controlling, at the filter control the adaptive filter, wherein the filtering of the output of the adaptive filter at the first filter is according to a function of the ratio $C=A_{wf}/B_{vm}$, where

A_{wf} is the ratio between a primary noise field at the microphone and a primary noise field at a virtual microphone, and B_{vm} is the ratio between an acoustic path from the loudspeaker to the microphone and an acoustic path from the loudspeaker to a virtual microphone.

Owing to the invention, it is possible to provide a signal processing system that will provide improved active noise cancellation at a user's ear, in an environment such as a vehicle. The cancellation provided by the system works on the virtual microphone idea, where the virtual microphone is estimated to be where the user's ear is located. In addition to using the difference in the acoustic path from the loudspeaker to the real microphone and from the loudspeaker to the virtual microphone, the difference in the acoustic (noise) field at the real and virtual microphones is also used (through the function C). These measures of the acoustic paths and noise fields are measured before the system is actually used, with a real microphone placed at the location of the (eventual) virtual microphone. It should be understood that these measured values are estimates, and the system operates using these estimates of the acoustic paths and acoustic fields.

In a first embodiment, the system further comprises a third filter interposed between the adaptive filter and the loudspeaker, the third filter arranged to filter the output of the adaptive filter prior to receipt by the loudspeaker. The third filter is arranged to filter according to the function C, where $C=A_{wf}/B_{vm}$. The first filter is arranged to filter according to the function $[(1-C).H_{em}]$, where $C=A_{wf}/B_{vm}$, and H_{em} is the acoustic path from the loudspeaker to the microphone. The second filter is arranged to filter according to the function H_{em}, where, again, H_{em} is the acoustic path from the loudspeaker to the microphone.

In a second embodiment, the first filter is arranged to filter according to the function $[(1-C)/C.H_{em}]$, where $C=A_{wf}/B_{vm}$, and H_{em} is the acoustic path from the loudspeaker to the microphone. The second filter is arranged to filter according to the function H_{em}/C, where $C=A_{wf}/B_{vm}$, and H_{em} is the acoustic path from the loudspeaker to the microphone.

Embodiments of the present invention will now be described, by way of example only, with reference to the accompanying drawings, in which:

Fig. 1 is a schematic diagram of a prior art active noise cancellation system,

Fig. 2 is a schematic diagram of a second, different, prior art active noise cancellation system,

Fig. 3 is a schematic diagram of a first embodiment of a signal processing system for active sound control,

Fig. 4 is a schematic diagram of a second embodiment of the signal processing system for active sound control, and

Fig. 5 is a flow diagram of a method of operating the signal processing system for active sound control.

Figure 3 shows a schematic diagram of a signal processing system that can be used for active sound control. The system comprises a reference signal generator R arranged to output a reference signal. The reference signal, which should have a high coherence with the actual acoustical noise, can be a signal from any sensor (accelerometer, microphone or directly from a tachometer) or derived (synthesized) from such a signal. It is not always an audio signal. In the case of multi-tonal engine cancellation, the block (R) represents a reference synthesizer.

An adaptive filter W_{fx} is arranged to receive the output of the reference signal generator R, with a loudspeaker 12 and a first filter 14 is also arranged to receive an output of the adaptive filter W_{fx}. The loudspeaker 12 is outputting an audio signal that will cancel the noise in the vehicle cabin at (or as near as possible) to the driver's ear. The virtual microphone 22 in Figure 3 is representing the assumed position of the driver's ear.

The system of Figure 3 also includes a real microphone 16, which picks up the output of the loudspeaker 12, via the acoustic path H_{em}, and the environmental noise. In addition, a summer 18 is arranged to receive outputs of the microphone 16 and the first filter 14. A second filter 20 is arranged to receive an output of the reference signal generator R (the reference signal), and a filter control LMS is arranged to receive outputs of the second filter 20 and the summer 18. The filter control LMS is arranged to control the adaptive filter W_{fx}, on the basis of the inputs received from the summer 18 and the filter 20.

In this embodiment of the invention, the system also further comprises a third filter C interposed between the adaptive filter W_{fx} and the loudspeaker 12, the third filter C arranged to filter the output of the adaptive filter W_{fx} prior to receipt by the loudspeaker 12. The third filter C is arranged to filter its input according to the function C, where $C=A_{wf}/B_{vm}$ (this ratio is discussed in more detail below).

The first filter 14 is arranged to filter the output of the adaptive filter W_{fx} according to a function of the ratio $C=A_{wf}/B_{vm}$, where A_{wf} is the ratio between a primary noise field at the microphone 16 and a primary noise field at a virtual microphone 22, and B_{vm} is the ratio between an acoustic path from the loudspeaker 12 to the microphone 16 and an acoustic path from the loudspeaker 12 to a virtual microphone 22. To be more precise, the function of C that is used by the first filter 14 is the function $[(1\text{-}C).H_{em}]$, where $C=A_{wf}/B_{vm}$, and H_{em} is the acoustic path from the loudspeaker 12 to the microphone 16. The second filter 20 is arranged to filter according to the function H_{em}, again, where H_{em} is the acoustic path from the loudspeaker (12) to the microphone (16).

To arrive at the functional blocks shown in Figure 3, it is assumed that the relation between the transfer function of the secondary paths, respectively from loudspeaker to error microphone (H_{em}) and virtual microphone (H_{vm}) can be expressed in a linear way:

$$H_{vm}(\omega) = B_{vm}(\omega)\, H_{em}(\omega) \qquad (1)$$

Where B_{vm} is the complex relationship between these two acoustic paths. It is also assumed that a similar relationship (A_{wf}) exists between the primary noise field at the error microphone (N_{em}) and at the virtual microphone (N_{vm}). This is expressed as:

$$N_{vm}(\omega) = A_{wf}(\omega)\, N_{em}(\omega) \qquad (2)$$

In the continuing discussion of the functions of these variables, for simplicity, the frequency dependency (ω) is dropped in the subsequent equations. If there is perfect cancellation of acoustic noise at the error microphone M_{em}, then this can be represented as follows:

$$R\, W_{fx}\, H_{em} = -N_{em} \qquad (3)$$

This equation can be explained as showing that the filtered (W_{fx}) reference signal after the secondary path (H_{em}) is equal to the primary noise field (except for the change in sign) at the error microphone. Similarly for the virtual microphone M_{vm} we have:

$$M_{vm} = N_{vm} + R\, W_{fx}\, H_{vm} \qquad (4)$$

using equation (2) $= A_{wf}\, N_{em} + R\, W_{fx}\, H_{vm}$ (5)

using equation (1) $= A_{wf}\, N_{em} + R\, W_{fx}\, (B_{vm}\, H_{em})$ (6)

rearranging this $= A_{wf}\, N_{em} + (R\, W_{fx}\, H_{em})\, B_{vm}$ (7)

using equation (3) $= A_{wf}\, N_{em} + (-N_{em})\, B_{vm}$ (8)

rearranging this $= N_{em}\, (A_{wf} - B_{vm})$ (9)

This shows that the signal at the virtual microphone can be expressed in terms of the primary noise field at the error microphone and the complex relationships between the noise fields and the transfer functions. When there is introduced an extra filter C after the FX-LMS filter, the equation for M_{vm} becomes in this case:

$$M_{vm} = N_{vm} + R\ W_{fx}\ C\ H_{vm} \quad (10)$$

$$= N_{em}\ (A_{wf} - C\ B_{vm}) \quad (11)$$

For a perfect cancellation at the virtual microphone M_{vm} would be equal to zero. In order to ensure that equation (11) equals zero:

$$N_{em} = 0 \quad (12)$$

or

$$C = A_{wf} / B_{vm} \quad (13)$$

The first solution (equation 12) is not of interest, because when there is no primary noise at all, it is not necessary to cancel that noise. So, in general with an extra filter C perfect cancellation can be achieved at a position different from that of the error microphone. This filter depends on the ratio of the difference in transfer function (B_{vm}) and the difference in wave field (A_{wf}), as shown by equation 13.

The introduction of the filter C into the embodiment shown in Figure 3 changes the structure of the output of the loudspeaker 12 and therefore adaptations need to be made elsewhere to take this into account. For a correct adaptation of the FX-LMS filters W_{fx}, the error signal E_{up} (traveling back to the LMS from the summer 18, should stay equal to the situation (without a virtual microphone). To achieve this, there is introduced a correction bridge H_b (the filter 14) from the output of the adaptive filter (W_{fx}) to the error microphone input signal.

$$E_{up} = N_{em} + R\ W_{fx}\ H_{em} = M_{em} + H_b \quad (14)$$

The effect of the bridge should be equal to:

$$H_b = R\ W_{fx}\ H_{em} + N_{em} - M_{em} \quad (15)$$

$$= R\ W_{fx}\ H_{em} + N_{em} - \quad R\ W_{fx} \quad C - N_{em} \quad (16)$$

$$= (1 - C)\ R\ W_{fx}\ H_{em} \quad (17)$$

In practical terms, it is only possible to apply an estimate of H_{em} for the bridge H_b. The output of the adaptive filter W_{fx} can be taken and filtered in the following way (using estimates of the paths H):

$$H_{b,filt} = (1-C)\ H_{em} \tag{18}$$

$$H_b = H_{b,filt}\ R\ W_{fx} \tag{19}$$

C has to be calculated before the system can be operated from the differences in noise field and acoustic paths. Figure 3 shows the first embodiment of the noise cancellation system, with the filter 14 working according to the equation 18 above, and the filter 20 filtering the reference signal according to an estimation of the path H_{em} from the loudspeaker 12 to the microphone 16.

A second embodiment of the system is shown in Figure 4. In this embodiment, the filter C is removed from the primary branch and moved to the bridge (the filter 14). The bridge filter $H_{b,filt}$ becomes:

$$H_{b,filt} = (1-C)/C\ H_{em} \tag{20}$$

The algorithm, of the embodiment of Figure 4, is still in principal the same as that shown in Figure 3, in terms of the actual output of the loudspeaker 12 and the actual noise cancellation taking place.

In operating the system, it could be assumed that ($A_{wf} = 1$). This means that the noise field is equal at both the error microphone and the virtual microphone. The bridge filter $H_{b,filt}$ becomes:

$$H_{b,filt} \quad = (1-C)/C\ H_{em} \tag{21}$$

$$= (1- (1/B_{vm}))/(1/B_{vm})H_{em} \tag{22}$$

$$= (B_{vm}-1)H_{em} = H_{vm} - H_{em} \tag{23}$$

For the reference filtering (filter 20):

$$H_{em}/C \quad = B_{vm}/A_{wf}\ H_{em} \tag{24}$$

$$= B_{vm}/H_{em} \tag{25}$$

$$= H_{vm} \tag{26}$$

In a practical embodiment of the system in a vehicle the system will operate with two virtual microphones (one for each ear of the driver) and the system will use two loudspeakers. The estimation of the virtual acoustic paths has to take place offline. The path from the loudspeaker (B_{vm}) to the virtual microphone can be obtained using a broadband white noise signal. The estimation of A_{wf} (the ratio of the noise fields at the microphones) is more difficult. This term has also to be estimated beforehand, prior to operation of the system. One method of estimating this value is to sweep the tacho signal by slowly pressing the gas pedal of the car, record the tacho signal and the noise at all of the microphones. The proposed method is suited for local ANC applications in car cabins. The invention can also be applied to the concept of personal sound spaces and extended to broadband ANC. The system supports a method for improving the performance of an Active Noise Cancelling (ANC) system by reducing the engine noise at the driver's ears. The algorithm involves the concept of virtual microphones and does not rely on the assumption that the sound pressure at the microphones is equal. The Active Noise Control can be an option for comfort and safety improvement within the car cabin. The system provides the active suppression of engine noise at the driver's ears.

Figure 5 summarizes the method of operating the signal processing system of Figures 4 and 5, for the active sound control. The method comprises, firstly generating (step 510) the reference signal with the generator R, which is filtered (step 512), at the adaptive filter (W_{fx}). The filtered reference signal may then pass through (step 514) a further filter C which additionally filters the reference signal with the ratio C.

The filtered reference signal is outputted (step 516), at the loudspeaker 12. This audio signal is received (step 518), at the real microphone 16. The output of the adaptive filter W_{fx} is also received by the filter 14 which operates in parallel to the loudspeaker 12 and microphone 16. The filtering (step 520) of the output of the adaptive filter W_{fx} at the first filter 14 is according to a function of the ratio $C=A_{wf}/B_{vm}$, as discussed in detail above. In the embodiment of Figure 3, the function of C that is used is $(1-C)\ H_{em}$, where H_{em} is an estimation of the acoustic path from the loudspeaker 12 to the microphone 16. In the embodiment of Figure 4 the function of C used is $(1-C)/C\ H_{em}$.

The output of the filter 14 and the microphone 16 are combined (step 522), at the summer 18. The original reference signal is also filtered (step 524), at the second filter 20, using a function based on the acoustic path H_{em}. The output of this filter 20 and of the summer 18 are received, at the filter control LMS, which controls (step 526) the adaptive filter W_{fx}.

CLAIMS:

1. A signal processing system for active sound control comprising a reference signal generator (R), an adaptive filter (W_{fx}) arranged to receive an output of the reference signal generator (R), a loudspeaker (12) arranged to receive an output of the adaptive filter (W_{fx}), a first filter (14), also arranged to receive an output of the adaptive filter (W_{fx}), a microphone (16), a summer (18) arranged to receive outputs of the microphone (16) and the first filter (14), a second filter (20) arranged to receive an output of the reference signal generator (R), and a filter control (LMS) arranged to receive outputs of the second filter (20) and the summer (18), the filter control (LMS) arranged to control the adaptive filter (W_{fx}), wherein the first filter (14) is arranged to filter the output of the adaptive filter (W_{fx}) according to a function of the ratio $C=A_{wf}/B_{vm}$, where A_{wf} is the ratio between a primary noise field at the microphone (16) and a primary noise field at a virtual microphone (22), and B_{vm} is the ratio between an acoustic path from the loudspeaker (12) to the microphone (16) and an acoustic path from the loudspeaker (12) to a virtual microphone (22).

2. A system according to claim 1, and further comprising a third filter (C) interposed between the adaptive filter (W_{fx}) and the loudspeaker (12), the third filter (C) arranged to filter the output of the adaptive filter (W_{fx}) prior to receipt by the loudspeaker (12).

3. A system according to claim 2, wherein the third filter (C) is arranged to filter according to the function C, where $C=A_{wf}/B_{vm}$.

4. A system according to claim 3, wherein the first filter (14) is arranged to filter according to the function $[(1-C).H_{em}]$, where $C=A_{wf}/B_{vm}$, and H_{em} is the acoustic path from the loudspeaker (12) to the microphone (16).

5. A system according to claim 4, wherein the second filter (20) is arranged to filter according to the function H_{em}, where H_{em} is the acoustic path from the loudspeaker (12) to the microphone (16).

6. A system according to claim 1, wherein the first filter (14) is arranged to filter according to the function $[(1\text{-}C)/C.H_{em}]$, where $C=A_{wf}/B_{vm}$, and H_{em} is the acoustic path from the loudspeaker (12) to the microphone (16).

7. A system according to claim 6, wherein the second filter (20) is arranged to filter according to the function H_{em}/C, where $C=A_{wf}/B_{vm}$, and H_{em} is the acoustic path from the loudspeaker (12) to the microphone (16).

8. A method of operating a signal processing system for active sound control comprising;

generating a reference signal,

filtering, at an adaptive filter (W_{fx}), the reference signal,

outputting, at a loudspeaker (12), the output of the adaptive filter (W_{fx}),

filtering, at a first filter (14), the output of the adaptive filter (W_{fx}),

receiving, at a microphone (16), the output of the loudspeaker (12),

combining, at a summer (18), the outputs of the microphone (16) and the first filter (14),

filtering, at a second filter (20), the reference signal,

receiving, at a filter control (LMS), the outputs of the second filter (20) and the summer (18), and

controlling, at the filter control (LMS) the adaptive filter (W_{fx}),

wherein the filtering of the output of the adaptive filter (W_{fx}) at the first filter (14) is according to a function of the ratio $C=A_{wf}/B_{vm}$, where A_{wf} is the ratio between a primary noise field at the microphone (16) and a primary noise field at a virtual microphone (22), and B_{vm} is the ratio between an acoustic path from the loudspeaker (12) to the microphone (16) and an acoustic path from the loudspeaker (12) to a virtual microphone (22).

9. A method according to claim 8, and further comprising filtering the output of the adaptive filter (W_{fx}) prior to receipt by the loudspeaker (12), at a third filter (C) interposed between the adaptive filter (W_{fx}) and the loudspeaker (12).

10. A method according to claim 9, wherein the filtering at the third filter (C) is according to the function C, where $C=A_{wf}/B_{vm}$.

11. A method according to claim 10, wherein the filtering at the first filter (14) is according to the function $[(1-C).H_{em}]$, where $C=A_{wf}/B_{vm}$, and H_{em} is the acoustic path from the loudspeaker (12) to the microphone (16).

12. A method according to claim 11, wherein the filtering at the second filter (20) is according to the function H_{em}, where H_{em} is the acoustic path from the loudspeaker (12) to the microphone (16).

13. A method according to claim 8, wherein the filtering at the first filter (14) is according to the function $[(1-C)/C.H_{em}]$, where $C=A_{wf}/B_{vm}$, and H_{em} is the acoustic path from the loudspeaker (12) to the microphone (16).

14. A method according to claim 13, wherein the filtering at the second filter (20) is according to the function H_{em}/C, where $C=A_{wf}/B_{vm}$, and H_{em} is the acoustic path from the loudspeaker (12) to the microphone (16).

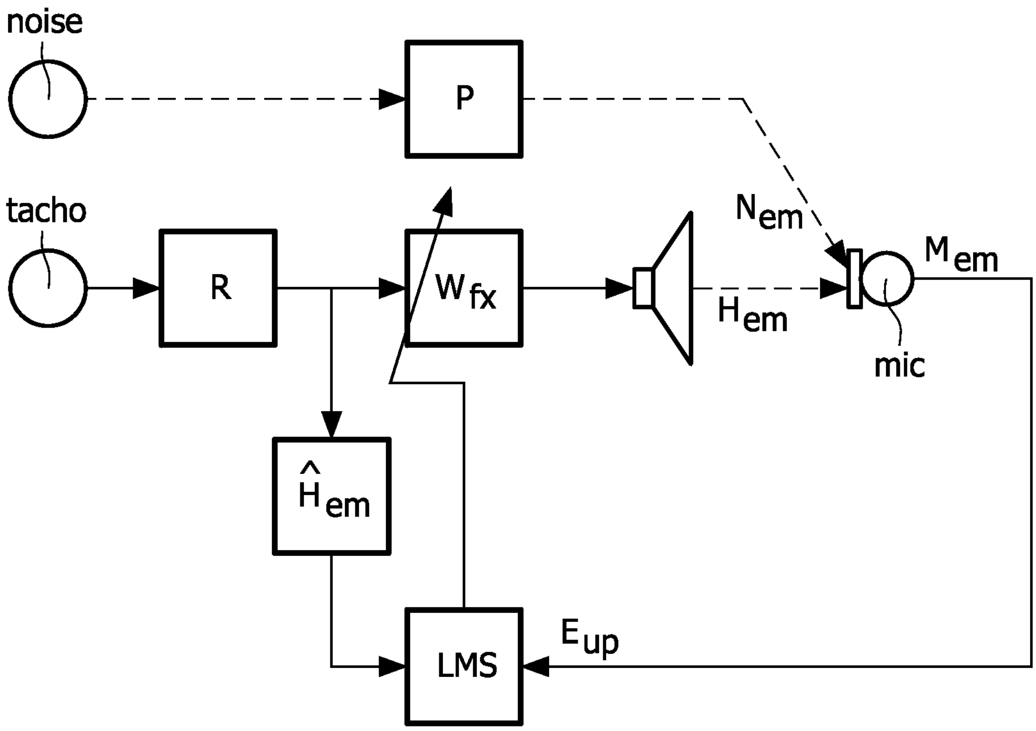


FIG. 1 PRIOR ART

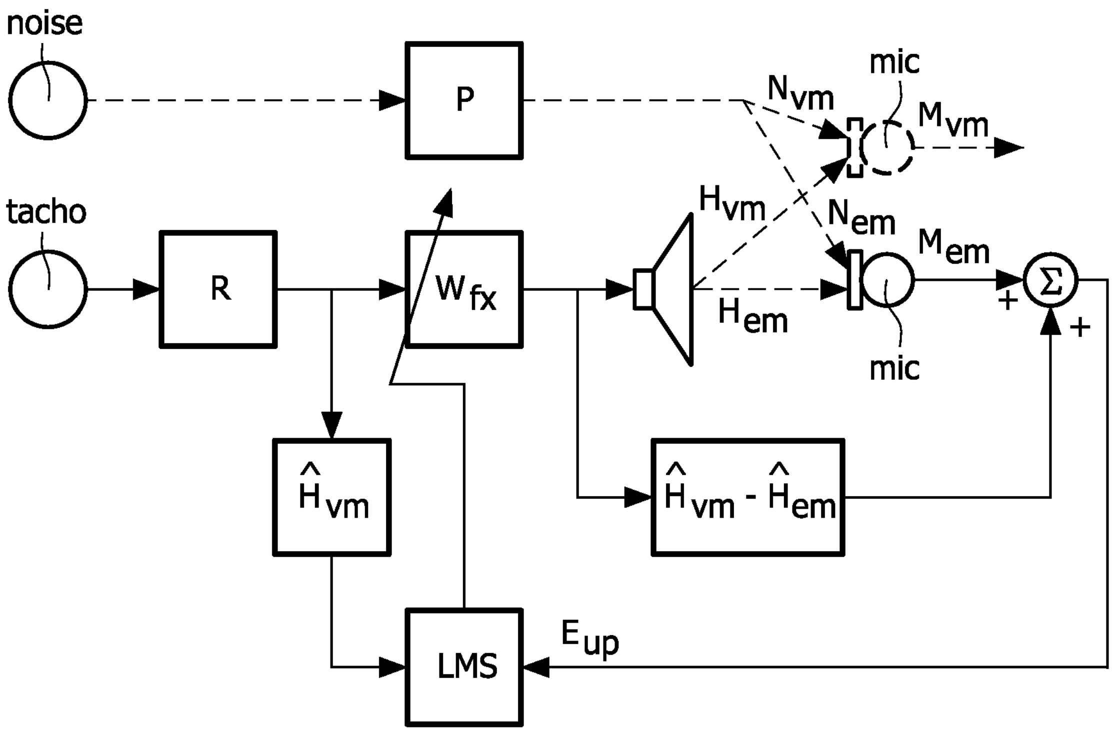
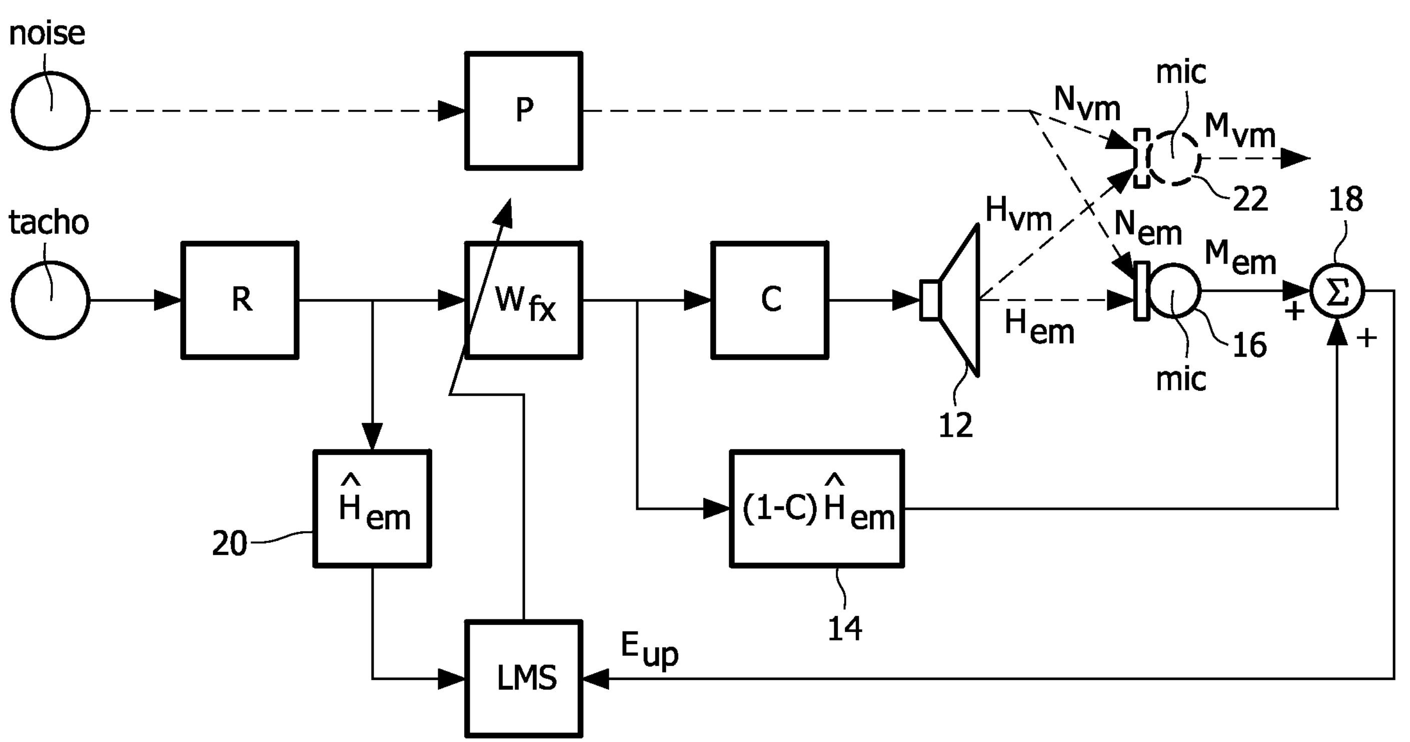


FIG. 2 PRIOR ART


noise
tacho
P
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C
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mic
Mvm
22
Hvm
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Mem
18
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mic
12
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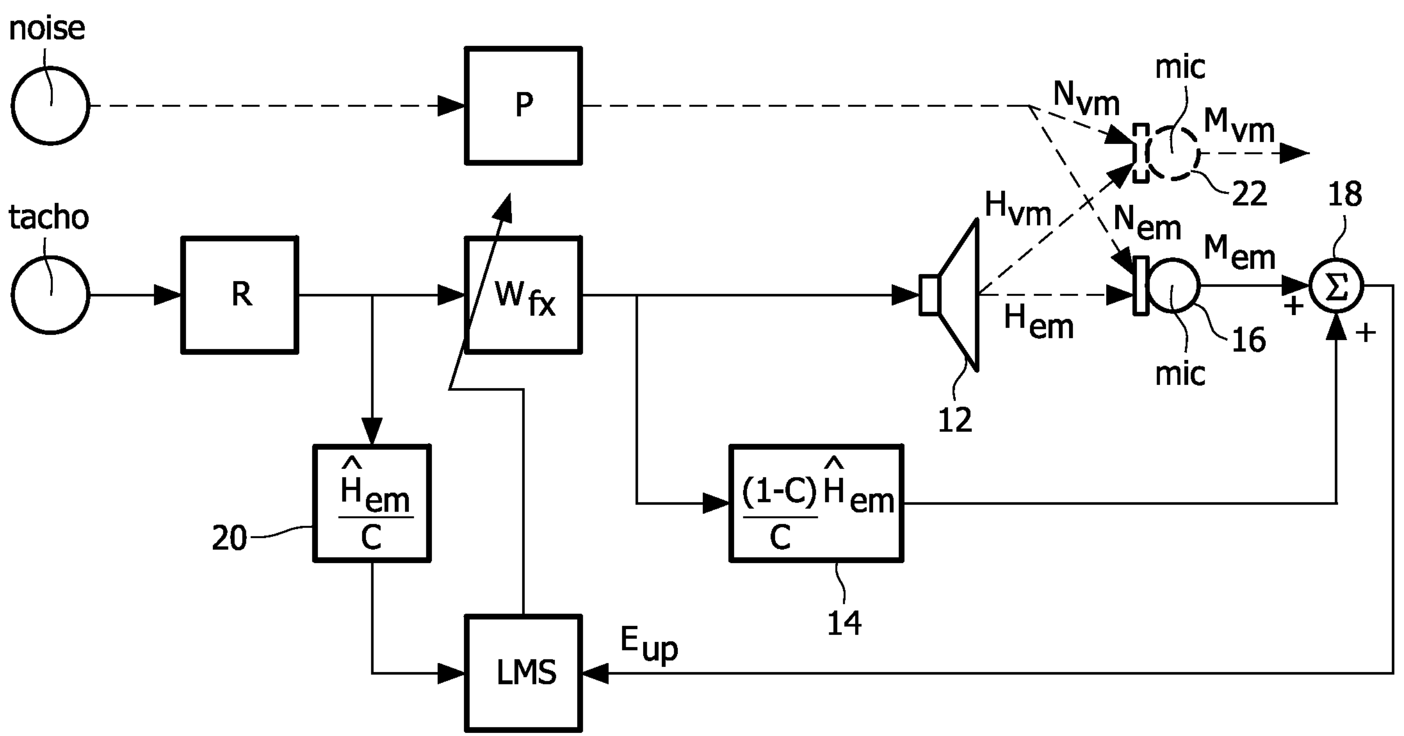
FIG. 3


noise
P
mic
Nvm
Mvm
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tacho
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Wfx
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C
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LMS
Eup

FIG. 4

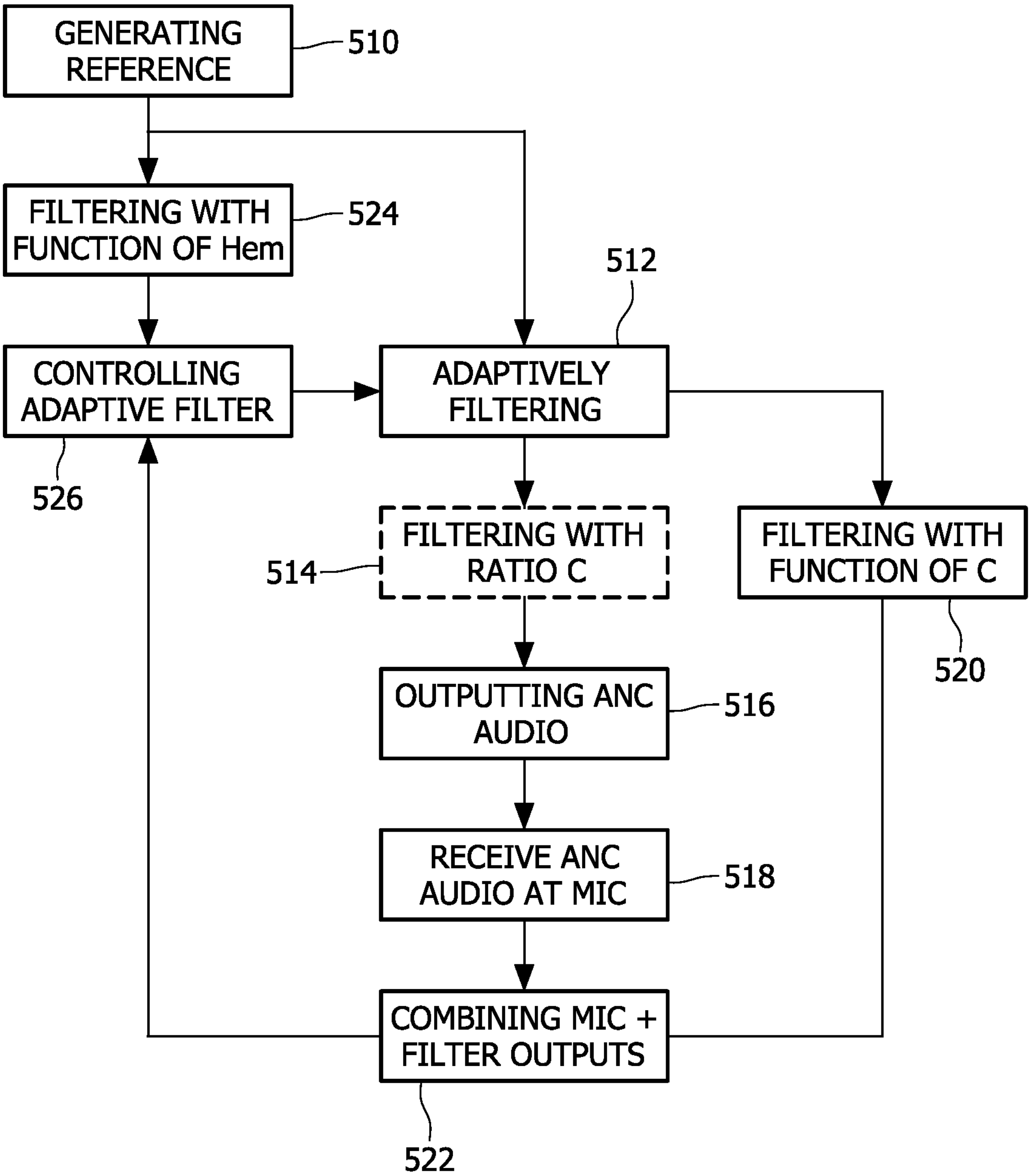
GENERATING REFERENCE
510
FILTERING WITH FUNCTION OF Hem
524
512
CONTROLLING ADAPTIVE FILTER
ADAPTIVELY FILTERING
526
FILTERING WITH RATIO C
514
FILTERING WITH FUNCTION OF C
520
OUTPUTTING ANC AUDIO
516
RECEIVE ANC AUDIO AT MIC
518
COMBINING MIC + FILTER OUTPUTS
522

FIG. 5

INTERNATIONAL SEARCH REPORT

International application No
PCT/IB2007/053531

A. CLASSIFICATION OF SUBJECT MATTER
INV. G10K11/178

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)
G10K 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, WPI Data

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US 5 701 350 A (POPOVICH STEVEN R [US]) 23 December 1997 (1997-12-23) column 1, line 1 - line 67; figure 1 column 3, line 42 - column 5, line 15; claim 1	1-14
A	DE 40 26 070 A1 (VOLKSWAGEN AG [DE]; GSP SPRACHTECHNOLOGIE GMBH [DE] VOLKSWAGEN AG [DE]) 28 February 1991 (1991-02-28) abstract; figure 1	1-14
A	US 5 381 485 A (ELLIOTT STEPHEN J [GB]) 10 January 1995 (1995-01-10) column 4, line 53 - column 6, line 30; figures 1-4A	1-14

-/--

[X] Further documents are listed in the continuation of Box C.

[X] 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

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"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.

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
23 January 2008	04/02/2008

Name and mailing address of the ISA/	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL – 2280 HV Rijswijk Tel. (+31–70) 340–2040, Tx. 31 651 epo nl, Fax: (+31–70) 340–3016	Righetti, Marco

C(Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	GARCIA-BONITO J ET AL: "GENERATION OF ZONES OF QUIET USING A VIRTUAL MICROPHONE ARRANGEMENT" JOURNAL OF THE ACOUSTICAL SOCIETY OF AMERICA, AIP / ACOUSTICAL SOCIETY OF AMERICA, MELVILLE, NY, US, vol. 101, no. 6, June 1997 (1997-06), pages 3498-3516, XP000696885 ISSN: 0001-4966 page 1, right-hand column - page 2, left-hand column -----	1-14

INTERNATIONAL SEARCH REPORT

Information on patent family members

International application No
PCT/IB2007/053531

Patent document cited in search report		Publication date	Patent family member(s)		Publication date
US 5701350	A	23-12-1997	NONE		
DE 4026070	A1	28-02-1991	GB	2236640 A	10-04-1991
US 5381485	A	10-01-1995	NONE		