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(71) Applicant: **Faurecia Creo AB**
582 16 Linköping (SE)

(72) Inventors:
- **Pignier, Nicolas Jean**
  116 31 Stockholm (SE)
- **Mattei, Christophe**
  587 23 Linköping (SE)
- **Risberg, Robert**
  589 37 Linköping (SE)

(74) Representative: **AWA Sweden AB**
Junkersgatan 1
582 35 Linköping (SE)

(54) **METHOD AND APPARATUS FOR SELECTING A SUBSET OF A PLURALITY OF INPUTS OF A MULTIPLE-INPUT-SINGLE-OUTPUT SYSTEM**

(57) A method for selecting a subset of a plurality of inputs of a Multiple-Input-Single-Output, MISO, system, the MISO system comprising: the plurality of inputs X_i, wherein i = 1, 2, ..., n, and a single output Y. The method comprises steps in a following order:
1) calculating a coherence value representing a coherence level between each of the plurality of inputs X_i and the single output Y;
2) among the plurality of inputs X_i, selecting an input having a largest coherence value;
3) creating a remaining group of inputs, wherein the remaining group of inputs consists all inputs of the plurality of inputs X_i except the previously selected input(s);
4) for each input of the remaining group of inputs, generating a corresponding conditioned input by conditioning the input;
5) for each conditioned input,
calculating a partial coherence value representing a coherence level between the conditioned input and the single output Y;
6) among the remaining group of inputs, selecting an input corresponding to a conditioned input having a largest partial coherence value.

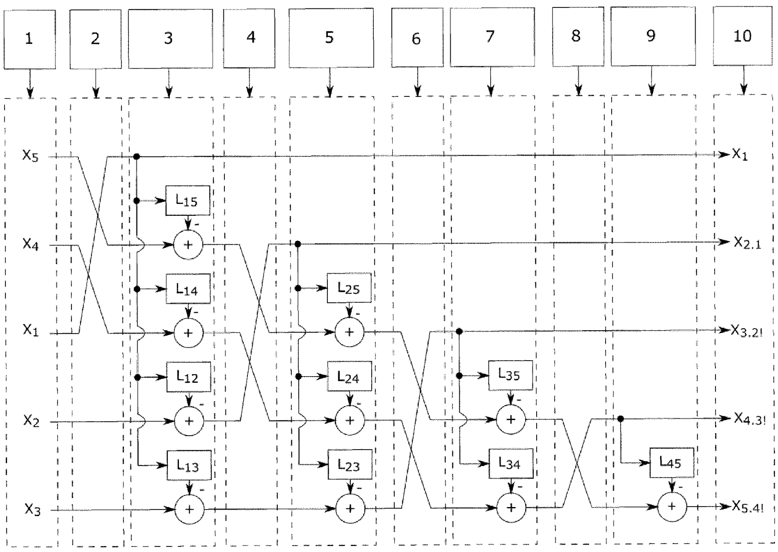

Fig 2

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Description

Technical field

[0001] The present document relates to a method and apparatus for selecting a subset of a plurality of inputs of a Multiple-Input-Single-Output, MISO, system.

Background

[0002] A MISO system may be used as a simplified model to describe a system comprising a plurality of inputs and a single output. The MISO system is suitable to characterise different systems, such as a MISO antenna system.

[0003] The MISO system may be represented by

$$\mathrm{Y} = \Sigma_{\mathrm{i}=1}^{\mathrm{L_x}} \mathrm{H_{iy}} \mathrm{X_i} + \mathrm{N}.$$

[0004] Wherein X_i, i = 1, 2, ..., i, ..., L_x, represents the plurality of inputs, and Y represents the single output. And H_{iy} is a transfer function for representing a linear relationship between each input X_i and output Y. N represents all possible deviations from an ideal model. That is, N represents everything that is not measured and accounted by the inputs.

[0005] Fig. 1a is an example of a MISO system. With the transfer function H_{iy}, a contribution Y_i, i = 1, 2, ..., i, ..., L_x of each input X_i to the output Y can be calculated. The output Y is a sum of each contribution Y_i and N.

[0006] In order to better characterise the MISO system, much studies have been done to identify the transfer function H_{iy}.

[0007] Bendat, J. S., & Piersol, A. G. (1980), Engineering applications of correlation and spectral analysis. New York, Wiley-Interscience, 1980. 315 p, Chap.10, has presented a method to identify H_{iy} in an optimal least-square manner using a recursive method. Here, H_{iy} does not necessarily represent the actual physically realizable characteristics of a given situation. Rather, H_{iy} are merely mathematical data processing results used for relating the single output Y to the plurality of inputs X_i by an optimum linear least-squares technique.

[0008] With Bendat's method, the MISO system of fig. 1a can be equivalently represented by an ordered set of conditioned inputs, wherein each input has been conditioned by the previous inputs, as shown in fig. 1b.

[0009] In fig. 1b, $X_{i\cdot(i-1)!}$ means a conditioned input X_i. That is, for each conditioned input X_i, linear effects of X_1 to X_{i-1} have been removed, e.g., by optimum linear least-squares prediction techniques. The conditioned inputs $X_{i\cdot(i-1)!}$ are uncorrelated to each other. The transfer functions L_{iy} of fig. 1b are in general different from the H_{iy} in fig. 1a.

[0010] The equivalent MISO system in fig. 1b may be represented by

$$Y = \Sigma_{i=1}^{L_x} L_{iy} X_{i.(i-1)!} + N$$

[0011] Thus, it is possible to determine H_{iy} and L_{iy} by the following relationships

$$H_{iy} = \frac{S_{iy.1,2,\ldots,(i-1),(i+1),\ldots,L_x}}{S_{ii.1,2,\ldots,(i-1),(i+1),\ldots,L_x}}$$

where S_{ij} is a cross-spectral density between a signal i and a signal j, and S_{ii} is an autospectrum of the signal i.

$$\begin{bmatrix} 1 & L_{12} & \cdots & L_{1L_x} \\ 0 & 1 & \ddots & \vdots \\ \vdots & \ddots & \ddots & L_{(L_x-1)L_x} \\ 0 & \cdots & 0 & 1 \end{bmatrix} \begin{bmatrix} H_{1y} \\ H_{2y} \\ \vdots \\ H_{L_x y} \end{bmatrix} = \begin{bmatrix} L_{1y} \\ L_{2y} \\ \vdots \\ L_{L_x y} \end{bmatrix}$$

[0012] Thus, it is possible to decompose the MISO system into uncorrelated subsystems each comprising a single input and allows in particular the computation of partial coherence γ^2 quantifying a degree of linearity between each of the plurality of inputs X_i and the output Y. Consequently, it is possible to determine which one(s) of the plurality of input X_i contribute most to the output, i.e. which one(s) of the plurality of input X_i is/are the most significant input(s).

[0013] Bendat's method for determining the H_{iy} and L_{1y} is recursive and depends on the order of the plurality of inputs X_i. That is, when the inputs of the same set of plurality of input X_i are arranged in a different order, with Bendat's method, different transfer functions H_{iy} and L_{1y} would be obtained.

[0014] Thus, with Bendat's method, the partial coherences γ^2 must be calculated for all possible orders of the plurality of inputs X_i for comparing the resulting partial coherences γ^2 between each of the plurality of inputs X_i and the output. This is a very complex and computationally costly process. A control unit with a high computational capacity is needed to handle such a high burden of calculations. Further, a quick reaction based on the resulted partial coherences γ^2 is impossible as the complex calculations will take time.

[0015] That is, Bendat's method is not suitable for characterising the MISO systems comprising an arbitrary order of the plurality of input signals, in terms of e.g. the partial coherence γ^2.

[0016] Hence, there is a need to provide an improved method for facilitating characterising a MISO system in a faster speed and by a reduced amount of calculation.

Summary

[0017] It is an object of the present disclosure, to provide a new method and apparatus for selecting a subset of a plurality of inputs of a Multiple-Input-Single-Output system, which eliminates or alleviates at least some of the disadvantages of the prior art.

[0018] The invention is defined by the appended independent claims. Embodiments are set forth in the appended dependent claims, and in the following description and drawings.

[0019] According to a first aspect, there is provided a method for selecting a subset of a plurality of inputs of a Multiple-Input-Single-Output, MISO, system, the MISO system comprising: the plurality of inputs X_i, wherein i = 1, 2, ...,n, and a single output Y. The method comprises steps in a following order:

1) calculating a coherence value representing a coherence level between each of the plurality of inputs X_i and the single output Y;
2) among the plurality of inputs X_i, selecting an input having a largest coherence value;
3) creating a remaining group of inputs, wherein the remaining group of inputs consists all inputs of the plurality of inputs X_i except the previously selected input(s);
4) for each input of the remaining group of inputs,

generating a corresponding conditioned input by conditioning the input;

5) for each conditioned input,
calculating a partial coherence value representing a coherence level between the conditioned input and the single output Y;
6) among the remaining group of inputs, selecting an input corresponding to a conditioned input having a largest partial coherence value.

[0020] The coherence level or partial coherence level is a mathematical way to represent a relationship between a plurality of signals or data sets. For example, it may be used to estimate a power transfer between an input and an output of a linear system. It may be used to estimate a contribution of an input to an output in the MISO system.

[0021] The multiple coherence γ^2 involving all inputs is independent on the order of the plurality of inputs X_i. However, partial coherences between each input and the output are dependent on the order of the plurality of inputs X_i.

[0022] It may be advantageous as a subset of the plurality of inputs of the MISO system may be selected, and optionally sorted, based on their respective contribution to the output, during recursive steps of an identification process for identifying one input among the plurality of inputs that has a largest coherence value or a largest partial coherence value.

[0023] By iterating the identification process, all inputs may be selected. Based on the selected order of the plurality of inputs, characterising a MISO system may be facilitated, as it is straightforward, e.g., to select an input which contributes the most or least among the plurality of inputs, to select a subset of an arbitrary number of inputs which contributes the most or least among the plurality of inputs, and to quantify a performance of any input, etc.

[0024] It may be advantageous as an initial order of the plurality of inputs can be random as it may not influence the resulted multiple coherences γ^2, which is different from Bendat's method.

[0025] Thus, comparing with Bendat's method, the amount of calculation may be reduced.

[0026] This may be advantageous as it may facilitate further processing based on the sorted plurality of inputs. For example, in order to save computation capacities, it is possible to only process the inputs having a coherence value larger than a threshold.

[0027] The MISO system may be defined in the context of an active noise control (ANC) system or an Active Road Noise Control (ARNC) system. The ANC/ARNC system typically involves i) one or several reference sensor(s) for detecting and/or measuring primary noises at noise sources; ii) one or several sound source(s), also known as secondary sound sources, e.g. loudspeakers of an existing audio system, for generating secondary noises to cancel the primary noises; iii) one or several error sensor(s) for detecting and/or measuring error signals representing a superposition of the primary noise and the secondary noise at different positions within an acoustic cavity, e.g., a vehicle cockpit; and iv) a control circuit, typically a digital signal processor (DSP) for performing an algorithm to generate control signals, such that the sound source(s) may be driven by the control signals to generate the secondary noises for cancelling the primary noises. The control signals are generated by filtering the reference signals generated by the reference sensor(s) with adaptive filters, which are updated by an adaptive algorithm, typically a least mean square (LMS) algorithm, to reduce a superposition of the primary and the secondary noises detected and/or measured by the error sensor(s), i.e. to reduce the error signal, or a squared pressure of a sound signal at the position of the error sensor(s).

[0028] The ANC and/or ARNC system and the adaptive algorithm are known and thus is not discussed in detail herein. The reference signals of the noise control systems may be considered as inputs of a MISO system. An acoustic signal measured at e.g., a location within an acoustic cavity, such as a position being close to an ear and/or head position of a driver within a car, may be considered as the output signal. Alternatively, the error signal may be may be considered as the output signal.

[0029] Determining the multiple coherence γ^2 may be useful in the ANC and/or ARNC systems. For example, a sound reduction that can be achieved by the ANC and/or ARNC system using L_x reference signals may be limited by the multiple coherence γ^2 quantifying the degree of linearity between each of the reference signals and a sound measured at a position. The relationship between the sound reduction ΔdB_{limit} and the multiple coherence γ^2 can be expressed as

$$\Delta \mathrm{dB}_{\mathrm{limit}} = 10 \log(1\text{-}\gamma^2).$$

[0030] Thus, with the inventive concept of the application, a subset of subband reference signals, and consequently the subset of the corresponding reference signals, and the subset of the corresponding reference sensors for generating the reference signals, which contribute most to a sound in a certain frequency range, may be determined. Further signal processings for noise reduction may involve only the selected subset of subband reference signals, and consequently the subset of the corresponding reference signals, and the subset of the corresponding reference sensors.

[0031] The relationship of one set A being a "subset" of another set B is also called inclusion or sometimes containment. The set A is a subset of the set B means that all elements of the set A are also elements of the set B. Thus, the set A is a subset of the set B even when the set A equals to the set B, i.e. the sets A and B consist exactly same elements. For example, if B={1, 3, 5} then A={1, 3, 5} is a subset of B.

[0032] A proper subset differs from the definition of subset. A proper subset C of the set B is a subset of the set B, which is not equal to the set B. In other words, if the set C is a proper subset of the set B, then all elements of the set C are elements of the set B. But the set B contains at least one element that is not an element of the set C. For example, if B={1,3,5}, then C={1} is a proper subset of B. But A={1, 3, 5} is not a proper subset of B.

[0033] The method may further comprise repeating the steps 3)- 6), until the remaining group of inputs consisting of a last one input, and selecting the last one input.

[0034] The method may be performed until a predetermined number of input being selected, e.g., 10% of the plurality of inputs.

[0035] The method may further comprise prior to selecting the last one input, performing steps 4) -5) for conditioning the last one input and calculating a partial coherence value between the conditioned last one input and the single output Y.

[0036] The method may be performed until all the inputs are selected. That is, the method may sort all the inputs based on a sequence of that each input of the plurality of inputs X_i is selected.

[0037] The method may further comprise repeating the steps 3)- 6), until the largest partial coherence value calculated at step 5) being smaller than a threshold.

[0038] The method may be performed until the calculated largest partial coherence value of the remaining group of inputs is less than a threshold. This may be advantageous as when the inputs of the remaining group of inputs comprise only the inputs contribute little to the output, the method may terminate rather than continuing iterations.

[0039] The method may further comprise sorting the plurality of inputs X_i based on a sequence that each input of the plurality of inputs X_i is selected, such that the input selected at the step 2) has a highest ranking.

[0040] That is, the input contributing most to the output may have a highest ranking, and the input contributing least to the output may have a lowest ranking.

[0041] The selected subset of the plurality of inputs X_i may comprise at least the input selected at the step 2).

[0042] A relationship between the conditioned inputs and the single output Y may be

$$Y = \Sigma_{i=1}^{n} L_{iy} X_{i.(i-1)!} + N,$$

wherein $X_{i.(i-1)!}$ may refer to an input X_i conditioned by the previously selected input(s) $X_{(i-1)!}$,
L_{iy} may refer to a transfer function, and
N may be a constant.

[0043] The step 4) of generating the conditioned inputs $X_{i.(i-1)!}$ may comprise conditioning each input of the remaining group of inputs by a previously selected input according to

$$X_{i.(i-1)!} = X_{i.(i-2)!} - L_{(i-1)i}X_{(i-1)(i-2)!}.$$

[0044] The term "conditioning" may refer to manipulating or processing a signal in such a way that it meets requirements of a next stage for further processing.

[0045] By removing the redundancies between the subband reference signals, the subband reference signals may be arranged in an arbitrary order. That is, the order of the subband reference signals may not play any role in determining their contributions to the output signal.

[0046] The method may further comprise prior to the step 1) of calculating the coherence value, arranging the plurality of input X_i in an arbitrary order.

[0047] This may be advantageous as with the inventive concept, the order of the inputs of the MISO system is less important in characterising the MISO system.

[0048] The method may further comprise prior to the step 1) of calculating the coherence value, performing an optimal least-square identification on the plurality of inputs X_i and the single output Y.

[0049] The inputs and the output of the MISO system may be processed, e.g., by an optimal least-square identification prior to selecting a subset of inputs according to the inventive concept.

[0050] According to a second aspect, there is provided a noise controlling method, comprising generating a plurality of reference signals representing a plurality of primary noises; generating a secondary noise in response to a control signal, for cancelling the plurality of primary noises; generating an error signal representing a superposition of the plurality of primary noises and the secondary noise at a position. The method further comprises: generating the control signal for generating the secondary noise, by executing an adaptive subband filtering algorithm based on the plurality of reference signals and the error signal; wherein the step of generating the control signal comprises: decomposing the plurality of reference signals and the error signal into subband reference signals and a subband error signal, respectively, for each subband of a plurality of subbands; updating a plurality of subband adaptive filters for at least one subband of the plurality of subbands, based on a proper subset of the subband reference signals of the at least one subband and the subband error signal of the at least one subband; updating at least one fullband adaptive filter based on the updated plurality of subband adaptive filters; generating the control signal by filtering the plurality of reference signals by the updated at least one fullband adaptive filter. The method further comprises: selecting the proper subset of the subband reference signals of the at least one subband by the method for selecting a subset of a plurality of inputs; wherein the subband reference signals correspond to the plurality of input X_i of the method for selecting a subset of a plurality of inputs; and wherein the subband error signal corresponds to the single output Y of the method for selecting a subset of a plurality of inputs. Or, the method further comprises generating a sound signal representing a sound at a second position, and the sound signal corresponds to the single output Y of the method for selecting a subset of a plurality of inputs.

[0051] The term "decomposing" may refer to splitting a fullband signal into multiple subband signals. The multiple decomposed subband signals may be processed independently. The decomposition may be achieved via a filter bank comprising, e.g., a set of bandpass filters. The terms fullband and subband may be in terms of frequency bands, or frequency ranges. Thus, the fullband signal may be a signal of a large frequency range, and the subband signal may be a signal of a small frequency range, being an interval of the large frequency range.

[0052] An adaptive filter may be a system having a transfer function controlled by variable parameters and a means to adjust those parameters according to an optimization algorithm.

[0053] The term "fullband adaptive filter" may refer to an adaptive filter adjusting fullband signals according to an optimization algorithm. The fullband signals here may be the reference signal, the error signal and the control signal.

[0054] The term "subband adaptive filter" may refer to an adaptive filter adjusting subband signals according to an optimization algorithm. The subband signals here may be the subband reference signals and the subband error signals.

[0055] The method may be performed within a car, truck, train, airplane, and any other acoustic cavity.

[0056] At least one reference sensor may be provided for generating the reference signal representing the primary noise. The at least one reference sensor may be an accelerometer, a microphone, or a tachometer.

[0057] At least one sound source may be provided for generating the secondary noise in response to the control signal, for cancelling the primary noise. The at least one sound source may be a loudspeaker, or a vibrating panel. At least one error sensor may be provided for generating the error signal representing a superposition of the primary noise and the secondary noise at the position. The at least one error sensor may be a microphone.

[0058] The primary noise may be a road noise, a wind noise, or an engine noise.

[0059] The coherence value and the partial coherence value may be calculated for a frequency range corresponding to the at least one subband.

[0060] The noise controlling method may be implemented by a ANC/ARNC system executing a delay-less subband FXLMS algorithm.

[0061] According to a third aspect, there is provided an apparatus for selecting a subset of a plurality of inputs of a Multiple-Input-Single-Output, MISO, system, the MISO system comprising: the plurality of inputs X_i, wherein i = 1, 2, ... n, and a single output Y. The apparatus comprising a control circuit configured to perform following functions in a following order:

a coherence value calculation function configured to calculate a coherence value for each of the plurality of inputs X_i, representing a coherence level between each of the plurality of inputs X_i and the single output Y;
a coherence value selection function configured to among the plurality of inputs X_i, select an input having a largest coherence value;
a group creation function configured to create a remaining group of inputs, wherein the remaining group of inputs consists all inputs of the plurality of inputs X_i except the previously selected input(s);
a condition function configured to, for each input of the remaining group of inputs, generate a conditioned input by conditioning the input;
a partial coherence value calculation function configured to calculate a partial coherence value for each conditioned input, representing a coherence level between the conditioned input and the single output Y;
a partial coherence value selection function configured to among the remaining group of inputs, select an input corresponding to a conditioned input having a largest partial coherence value.

[0062] The control circuit may be any type of processor, e.g., a digital signal processor (DSP), or a central processor unit (CPU).

[0063] The control circuit may be configured to perform an iteration function configured to repeat the group creation function, the condition function, the partial coherence value calculation function, and the partial coherence value selection function, until the remaining group of inputs consisting of a last one input, and to select the last one input, or until the largest partial coherence value calculated by the partial coherence value calculation function being smaller than a threshold.

[0064] The control circuit may be configured to perform a sorting function configured to sort the plurality of inputs X_i based on a sequence that each input of the plurality of inputs X_i is selected, such that the input selected by performing the coherence value selection function has a highest ranking.

[0065] According to a fourth aspect, there is provided a noise controlling system, comprising: a plurality of reference sensors configured to generate a plurality of reference signals representing a plurality of primary noises, respectively; a sound source configured to generate a secondary noise in response to a control signal, for cancelling the plurality of primary noises; an error sensor configured to generate an error signal representing a superposition of the plurality of primary noises and the secondary noise at a position; and a control unit configured to generate the control signal by executing an adaptive subband filtering algorithm, based on the plurality of reference signals and the error signal. The control unit is further configured to: decompose the plurality of reference signals and the error signal into subband reference signals and a subband error signal, respectively, for each subband of a plurality of subbands; update a plurality of subband adaptive filters for at least one subband of the plurality of subbands, based on a proper subset of the subband reference signals of the at least one subband and the subband error signal of the at least one subband; update at least one fullband adaptive filter based on the updated plurality of subband adaptive filters; generate the control signal by filtering the plurality of reference signals by the updated at least one fullband adaptive filter. The noise controlling system further comprises the apparatus for selecting a subset of a plurality of inputs, for selecting the proper subset of the subband referent signals; and wherein the subband reference signals correspond to the plurality of input X_i of the apparatus for selecting a subset of a plurality of inputs; and wherein the subband error signal corresponds to the single output Y of the apparatus for selecting a subset of a plurality of inputs. Or, the noise controlling system further comprises a sensor configured to generate a sound signal representing a sound at a second position, and the sound signal corresponds to the single output Y of the apparatus for selecting a subset of a plurality of inputs.

[0066] According to a fifth aspect, there is provided a non-transitory computer readable recording medium having computer readable program code recorded thereon which when executed on a device having processing capability is configured to perform the method for selecting a subset of a plurality of inputs.

Brief Description of the Drawings

[0067]

Fig. 1a illustrates an example of a MISO system.
Fig. 1b illustrates an equivalent of the MISO system of fig. 1a.
Fig. 2 illustrates a procedure for sorting a plurality of inputs.
Fig. 3 illustrates an example of a delay-less subband FXLMS algorithm implemented in an ANC or ARNC system.
Fig. 4 illustrates an example of a noise controlling system.
Fig. 5 illustrates an example of a noise controlling system.
Fig. 6 illustrates four diagrams of noise reduction measurements.
Fig. 7 illustrates four diagrams of noise reduction measurements.
Figs 8a-8c illustrate diagrams of measured SPL values.
Figs 9a-9c illustrate diagrams of measured SPL values.
Fig. 10 illustrates a diagram of measured SPL values.
Fig. 11 illustrates an example of a function $\chi^{(k)}$.
Fig. 12 illustrates a diagram of numbers of multiplications per sample of different ANC methods.

Description of Embodiments

[0068] In connection to fig. 2, the method for selecting a subset of a plurality of inputs of the MISO system of figs 1a-1b, is discussed in detail.

[0069] The MISO system of fig. 2 has five inputs X_1 to X_5 and one single output (not shown). However, the number of the inputs can be any positive integer.

[0070] In fig. 2, the inputs X_1 to X_5 are numbered according to a final order of inputs sorted based on their respective coherence to the output, i.e. their respective contribution to the output, e.g. in a frequency range of interest. That is, in fig. 2, the input X_1 has a largest coherence to the output and the input X_5 has a least coherence to the output. This is only to simplify the illustration. However, the inputs may be arranged in any order.

[0071] Step 1. The inputs X_1 to X_5 may be arranged in an arbitrary order. In fig. 2, the inputs are arranged in this order: X_5, X_4, X_1, X_2, X_3.

[0072] Step 2. A coherence value between each of the inputs X_1 to X_5 and the output is calculated. The input having a largest coherence value in the frequency range of interest, is selected as a first input. Here, X_1 is the first input, which has the largest coherence value. The remaining group of inputs consists inputs X_1 to X_5.

[0073] Step 3. A first stage of the MISO system identification is performed, which essentially consists of conditioning the remaining group of inputs by the input selected in the step immediately prior to the present step 3, i.e. X_1 selected in step 2. The conditioning step may remove linear contributions L_{15},L_{14}, L_{12}, L_{13} of the inputs X_5, X_4, X_2 and X_3 to the output, respectively, which has already been accounted for the selected input X_1.

[0074] The conditioned inputs $X_{5.1}$,$X_{4.1}$,$X_{2.1}$,$X_{3.1}$ in this example may be calculated as:

$$X_{j.1} = X_j - L_{1j}X_1, \quad j = 2,3,4,5$$

[0075] Step 4. A partial coherence $\gamma^2_{\mathrm{iy}.1}$, i= 2, ..., 5, between each of the conditioned input and the output is calculated. The input corresponding to a conditioned input having a largest partial coherence value in the frequency range of interest is selected as a second input. Here, X_2 is the second input, which has the largest partial coherence value among the inputs X_2 to X_5. Now the remaining group of inputs consists inputs X_3 to X_5.

[0076] Step 5. A second stage of the MISO system identification is performed, which essentially consists of conditioning the remaining group of inputs by the input selected in the step immediately prior to the present step 5, i.e. X_2 selected at step 4. The conditioning step may remove linear contributions L_{25},L_{24}, L_{23} of the inputs X_5, X_4 and X_3 to the output, respectively, which has already been accounted for the selected input X_2.

[0077] The conditioned inputs $X_{5.2!}$,$X_{4.2!}$, $X_{3.2!}$ in this example may be calculated as:

$$X_{j.2!} = X_{j.1} - L_{2j}X_{2.1}, \quad j = 3,4,5$$

[0078] From and including the step 5, an iteration of the steps 3-4 can be performed, e.g., the steps 5-6 and the steps 7-8 are iterations of the steps 3-4. By iteration, the remaining group of inputs are conditioned by the previously selected input, then a partial coherence between each conditioned input and the output is calculated, and one input corresponding to the conditioned input having a largest partial coherence value in the frequency range of interest among the remaining group of input is selected. The iteration may continue until the remaining group of inputs consists of a last one input, X_5.

[0079] However, the method does not need to be performed until all the inputs are selected, as in fig. 2. For example, if only a subset of all the input(s) that contribute most to the output is to be identified, it is sufficient to perform the method until a sufficient number of the inputs are selected. That is, the iteration may continue until a predetermined number of inputs have been selected. For example, only top three inputs having largest coherences are to be selected.

[0080] Alternatively, it is possible to determine a threshold, e.g., a minimal partial coherence value. The method can be performed until a largest partial coherence value of the remaining group of inputs is equal to or below the threshold.

[0081] Step 9. A last stage of the MISO system identification is performed, which essentially consists of conditioning the last one input X_5 by the input selected in the step immediately prior to the present step 9, i.e. X_4 selected at step 8. The conditioning step may remove a linear contribution L_{45} of the last one input X_5 to the output, which has already been accounted for the selected input X_4.

[0082] The conditioned input $X_{5.4!}$ in this example may be computed as:

$$X_{5.4!} = X_{5.3!} - L_{45}X_{4.3!}$$

[0083] Step 10. The last one input X_5 is selected as the last input, i.e. the fifth input in fig. 2. The plurality of inputs may be sorted based on a sequence that each of the plurality of inputs is selected. Further, the plurality of inputs has been conditioned by the previously selected inputs. The first selected input, here X_1, is not conditioned as no input has been previously selected.

[0084] The method may be performed with an arbitrary number of inputs. The inputs may have an arbitrary initial order. Once the method is performed until all the inputs are selected, as shown in fig. 2, all the inputs may be sorted by a decreased coherence, i.e. a decreased contribution, to the output. Since all the inputs, except the first selected input, are conditioned by the previously selected inputs, redundancy of information between the inputs may be reduced. That is, each input selected in the system at steps 2, 4, 6 and 8 maximally contributes to the output in terms of an added (non-redundant) information.

[0085] Thus, the method may select, and optionally sort, the inputs, e.g., in steps 2, 4, 6, 8 and 10 of fig. 2, before the remaining group of inputs being used in a next iteration.

[0086] The method for selecting at least one input from a plurality of inputs of a MISO system, may be implemented in many existing MISO system, or systems comprising a MISO subsystem, such as an ANC or ARNC system of fig. 3.

[0087] Fig. 3 illustrates an example of a known delay-less subband FXLMS algorithm implemented in an ANC or ARNC system.

[0088] The method shown in fig. 3 can be found in, for example, [1] Cheer, J., & Daley, S. (2017), An investigation of delayless subband adaptive filtering for multi-input multi-output active noise control applications. IEEE/ACM Transactions on Audio, Speech, and Language Processing, 25(2), 359-373; and [2] Milani, A. A., Panahi, I. M., & Loizou, P. C. (2009), A new delayless subband adaptive filtering algorithm for active noise control systems. IEEE transactions on audio, speech, and language processing, 17(5), 1038-1045.

[0089] In fig. 3, the system comprises L_x reference sensors 1, L_y sound sources 3, L_e error sensors 4 and an adaptive subband filtering algorithm 8 executed by a control circuit (not shown). The L_x reference sensors 1 generate L_x reference signals x(n), respectively, which may be represented in a vector notation by $x(n) = (x_1(n), ...,x_{Lx}(n))$. The L_e error sensors 4 generate L_e error signals e(n), respectively, which may be represented in a vector notation by $e(n) = (e_1(n), ..., e_{L_e}(n))$. The L_y sound sources 3 are driven by L_y control signals y(n), respectively, which may be represented in a vector notation by notation by $y(n) = (y_1(n), ...,y_{L_y}(n))$.

[0090] The adaptive subband filtering algorithm 8 is used to generate the control signals y(n) by filtering the reference signals x(n) with adaptive filters W(n), such that the sound sources 3 may generate the secondary noises to cancel primary noise 9. The sound sources 3 and the error sensors 4 are provided in an acoustic propagation domain 12. The acoustic propagation domain 12 may be either open or closed.

[0091] The primary noise 9 may be a road noise, generated by e.g., an interaction of a vehicle with a road through wheels. The primary noise 9 may also be any other type of noises, such as a wind noise or an engine noise, provided that the noise may be characterised by physically measurable reference signals.

[0092] The adaptive filters W(n) may be updated according to any known method, such as a LMS algorithm, to reduce a superposition of the primary noise 9 and the secondary noise at the error sensors 4, i.e. to reduce the error signals e(n), or a squared pressure at the error sensors 4. The adaptive filters W(n) may be updated continuously. The adaptive subband filtering algorithm 8 in fig. 3 may be a delay-less subband FXLMS algorithm.

[0093] The reference signals $x(n) = (x_1(n), ...,x_{L_x}(n))$ may be first filtered by a secondary path model S 11, which represents a plurality of acoustic transmission paths from each of the plurality of sound sources, also known as secondary sound sources, to each of the plurality of error sensors. Thus, the number of secondary paths may be the number of the sound sources multiplied by the number of the error sensors, i.e. L_y*L_e in this example.

[0094] The filtered reference signals x'(n) may be represented in a vector notation by $x'(n) = (x_{1,1,1}(n), \dots x_{Le,1,1}(n), \dots, x_{Le,Lx,1}(n), \dots, x_{Le,Lx,Ly}(n))$ The filtered reference signals x'(n) may be filtered by a filter bank 10. As a consequence, K subband signals may be generated from each of the filtered reference signals. The filter band 10 may comprise a decimation step of a factorD,wherein $\mathrm{D} = \frac{\mathrm{K}}{4}.$ This may result in L_x subband reference signals $x'^{(k)}$ per subband. Further, since the reference signals x(n) are filtered by the secondary paths model S 11, which comprises $L_y{*}L_e$ secondary paths, each subband reference signal $x'^{(k)}$ contains $L_y{*}L_e$ signals. This may result in a total $L_x{*}L_y{*}L_e$ subband reference signals $x'^{(k)}$ per subband.

[0095] A subband reference matrix $\boldsymbol{R}^{(k)}$ may be a matrix, wherein each coefficient is made of the subband filtered reference signals. The subband reference matrix $\boldsymbol{R}^{(k)}$ for the subband k may be expressed as

$$\boldsymbol{R}^{(k)} = \begin{bmatrix} \boldsymbol{r}_1^T(n) & \cdots & \boldsymbol{r}_1^T(n - ISAF + 1) \\ \vdots & \ddots & \vdots \\ \boldsymbol{r}_{L_e}^T(n) & \cdots & \boldsymbol{r}_{L_e}^T(n - ISAF + 1) \end{bmatrix},$$

wherein

$$\boldsymbol{r}_{l_e}^T(n - i + 1) = \left[x'_{l_e\boldsymbol{11}}(n - i + 1) \quad \dots \quad x'_{l_e\boldsymbol{1L_x}}(n - i + 1) \quad \dots \quad \dots \quad x'_{l_e\boldsymbol{L_yL_x}}(n - i + 1)\right],$$
$$i = 1 \dots ISAF.$$

[0096] *ISAF* may represent a length of the subband adaptive filters of the subband k. n may refer to a time step n. For example, when n refers to a current time step, n-1 refers to a previous time step and n+1 refers to a next time step.

[0097] The error signals e(n) are also decimated by the filter bank 10, as the reference signals x(n). This results in L_e subband error signals $e^{(k)}$ per subband.

[0098] Preferably, out of the K subbands created by the filter bank 10, only the first $\frac{\mathrm{K}}{2} + 1$ subbands are used. The others subbands may contain merely redundant information.

[0099] For each subband k of the first $\frac{\mathrm{K}}{2} + 1$ subbands, the subband adaptive filters $W^{(k)}$ are updated by an adaptive algorithm, such as an LMS algorithm as shown in fig. 3, based on the subband reference signals, the subband error signals. The control signals $\boldsymbol{y}(n)$ are generated from the fullband reference signals $\boldsymbol{x}(n)$, using the fullband adaptive filters.

[0100] When the subband adaptive filter(s) $W^{(k)}$ of each subband k, preferably each subband of the first $\frac{\mathrm{K}}{2} + 1$ subbands are updated, the fullband adaptive filters may be reconstructed, based on the updated subband adaptive filters by a well-known scheme 7, e.g., a weight or frequency stacking scheme.

[0101] The filter bank 10 may be an analysis filter bank, such as a Uniform Discrete Fourier Transform Modulated, UDFTM, filter bank. The weight stacking scheme may be a proposed Fast Fourier Transform weight stacking scheme described in [2].

[0102] For a subband in fig. 3, the plurality of subband reference signals may be considered to be the plurality of inputs of the MISO subsystem. An acoustic signal measured at e.g., a location within an acoustic cavity, such as a position being close to an ear and/or head position of a driver within a car, may be considered to be the output of the MISO subsystem. Alternatively, the error signal may be considered to be the output of the MISO subsystem. The frequency range of interest may correspond to a frequency range of the subband of fig. 3.

[0103] Thus, the method for selecting at least one input from a plurality of inputs of a MISO system may be implemented in the ANC/ARNC system to determine a subset of the subband reference signals. The subset of the subband reference signals may contribute most to the output signal. That is, it is possible to use only the subset of the subband reference signals for updating the subband adaptive filters for the subband. Each subband reference signal of the subset of the plurality of subband reference signals may have a significant contribution to the output signal.

[0104] The output signal may be generated by one of the existing error sensors. Alternatively, the output signal may be generated by an additional sensor, such as a monitor sensor, being typically a microphone placed at a listening position within the acoustic cavity. The listening position is usually in proximity of a head or an ear of a person within the acoustic cavity, such as a driver/passenger within a car. The method to determine the subsets of subband reference signals for a subband may started by performing an operational measurement of the reference and/or the error signal(s) in a typical operating condition. Then, a subset of the most significant subband reference signal(s) may be selected for updating the subband adaptive filters for each subband.

[0105] In connection with figs 4-5, a noise controlling system will be discussed in detail.

[0106] In fig. 4, for each subband k, preferably of the first $\frac{\mathrm{K}}{2}+1$ subbands, a proper subset of the filtered subband reference signals of the subband k may be selected, e.g., by a reference signal selecting unit 5 of the control circuit (not shown). Optionally, for each subband k, a proper subset of the subband error signals of the subband k may be selected, e.g., by an error signal selecting unit 6 of the control circuit.

[0107] For each subband k, each subband adaptive filter may correspond to a secondary sound source for generating the secondary noise. Optionally, for each subband k, a proper subset of the subband adaptive filters k may be selected, e.g., by an adaptive filter selecting function of the control circuit (not shown).

[0108] The reference signal selecting unit 5 may be provide for selecting a subset of the subband reference signals. The error signal selecting unit 6 may be provide for selecting a subset of the subband reference signals. An adaptive filter selecting function may be provided for selecting a subset of the subband adaptive filter for update.

[0109] For each subband k, only the selected subset of the subband reference signals, optionally the selected subset of the subband error signals are used for updating the subband adaptive filters $W^{(k)}$. The superscript $^{(k)}$ may represent a quantity related to the subband k.

[0110] The reference signal selecting unit 5 may comprise a function $\chi^{(k)}$ for selecting a subset, preferably a proper subset, of the L_x reference sensors 1 for the subband k. The function $\chi^{(k)}$ may be a function defining which of the L_x reference sensors are to be selected, and/or activated, for the subband k. The function $\chi^{(k)}$ may be predetermined. For the subband k, only the subband reference signals decomposed from the reference signals generated by the selected reference sensors are to be used. $\mathrm{L}_{\mathrm{x}}^{(\mathrm{k})}$ may be the number of the reference sensors selected for the subband k. $\mathrm{L}_{\mathrm{x}}^{(\mathrm{k})}$ may be smaller than L_x. That is, $\mathrm{L}_{\mathrm{x}}^{(\mathrm{k})} \leq \mathrm{L}_{\mathrm{x}}$.

[0111] The selected subset of subband reference signals used to update the selected subset of subband adaptive filters, e.g., defined by the function $\chi^{(k)}$, may be selected based on the physics properties of each subband, e.g., the different frequency ranges. For example, for a subband corresponding to a low frequency range, a subband reference signal decomposed from a reference signal generated by a reference sensor for detecting a high frequency noise may not be selected for updating the subband adaptive filters of this subband This may allow a reduced computational cost, in addition to a potential gain in performance.

[0112] The selected subset of the subband error signals $e^{(k)}$ for the subband k may be expresses as

$$\mathrm{e}^{(\mathrm{k})}(\mathrm{n})=\begin{bmatrix}\mathrm{e}_{\epsilon^{(\mathrm{k})}(1)}\\ \vdots \\ \mathrm{e}_{\epsilon^{(\mathrm{k})}\left(\mathrm{L}_{\mathrm{e}}^{(\mathrm{k})}\right)}\end{bmatrix}.$$

[0113] The subband reference matrix $R^{(k)}$ for the subband k may be expressed as

$$\boldsymbol{R}^{(k)}=\begin{bmatrix}\boldsymbol{r}^{\boldsymbol{T}}_{\epsilon^{(k)}(1)}(n) & \cdots & \boldsymbol{r}^{\boldsymbol{T}}_{\epsilon^{(k)}(1)}(n-ISAF+1)\\ \vdots & \ddots & \vdots \\ \boldsymbol{r}^{\boldsymbol{T}}_{\epsilon^{(k)}(L_e^{(k)})}(n) & \cdots & \boldsymbol{r}^{\boldsymbol{T}}_{\epsilon^{(k)}(L_e^{(k)})}(n-ISAF+1)\end{bmatrix},$$

wherein

$$\boldsymbol{r}^{\boldsymbol{T}}_{\epsilon^{(k)}(l_e)}(n-i+1) = \left[x'_{\epsilon^{(k)}(l_e)\boldsymbol{\psi}^{(\boldsymbol{k})}(\boldsymbol{1})\boldsymbol{\chi}^{(\boldsymbol{k})}(\boldsymbol{1})}(n-i+1) \quad \dots \quad x'_{\epsilon^{(k)}(l_e)\boldsymbol{\psi}^{(\boldsymbol{k})}(\boldsymbol{1})\boldsymbol{\chi}^{(\boldsymbol{k})}\left(\boldsymbol{L}_{\boldsymbol{x}}^{(\boldsymbol{k})}\right)}(n-i+1) \quad \dots \quad \dots \quad x'_{\epsilon^{(k)}(l_e)\boldsymbol{\psi}^{(\boldsymbol{k})}\left(\boldsymbol{L}_{\boldsymbol{y}}^{(\boldsymbol{k})}\right)\boldsymbol{\chi}^{(\boldsymbol{k})}\left(\boldsymbol{L}_{\boldsymbol{x}}^{(\boldsymbol{k})}\right)}(n-i+1)\right], \quad i = 1 \dots ISAF.$$

[0114] *ISAF* may represent a length of the subband adaptive filters of the subband k.

[0115] Each *x'* may represent a selected subband filtered reference signal, corresponding to the selected reference sensors $\chi^{(k)}(l_x)$, the selected error sensors $\varepsilon^{(k)}(l_e)$ and the selected sound sources $\psi^{(k)}(l_y)$.

$$x'_{\epsilon^{(k)}(l_e)\psi^{(k)}(l_y)\chi^{(k)}(l_x)}(n-i+1) = \widehat{\boldsymbol{g}}^{T}_{\epsilon^{(k)}(l_e)\boldsymbol{\psi}^{(k)}(\boldsymbol{l_y})}\boldsymbol{x}_{\boldsymbol{\chi}^{(k)}(\boldsymbol{l_x})}(n-i+1)$$

$$\widehat{\boldsymbol{g}}^{T}_{\epsilon^{(k)}(l_e)\boldsymbol{\psi}^{(k)}(\boldsymbol{l_y})} = \left[\hat{g}_{\epsilon^{(k)}(l_e)\boldsymbol{\psi}^{(k)}(\boldsymbol{l_y})}(1) \quad \cdots \quad \hat{g}_{\epsilon^{(k)}(l_e)\boldsymbol{\psi}^{(k)}(\boldsymbol{l_y})}(J)\right]$$

wherein J is a length of the filters for the secondary path model.

[0116] The secondary path models are finite impulse responses between the sound sources, i.e. the secondary sound sources, and the error sensors, mathematically represented by J coefficients.

[0117] For each subband k, the subband adaptive filters $W^{(k)}$ may be expressed as

$$\mathrm{W}^{(\mathrm{k})} = \begin{bmatrix} \mathrm{w}_1^{(\mathrm{k})} \\ \vdots \\ \mathrm{w}_{\mathrm{ISAF}}^{(\mathrm{k})} \end{bmatrix}.$$

[0118] Each subband adaptive filter $\mathrm{w}_{\mathrm{i}}^{(\mathrm{k})}$ of the subband adaptive filters $W^{(k)}$ may be expressed as

$$\mathrm{w}_{\mathrm{i}}^{(\mathrm{k})} = \left[\mathrm{w}^{(\mathrm{k})}_{\psi^{(\mathrm{k})}(1)\chi^{(\mathrm{k})}(1)\mathrm{i}} \quad \cdots \quad \mathrm{w}^{(\mathrm{k})}_{\psi^{(\mathrm{k})}(1)\chi^{(\mathrm{k})}\left(\mathrm{L}_{\mathrm{x}}^{(\mathrm{k})}\right)\mathrm{i}} \quad \mathrm{w}^{(\mathrm{k})}_{\psi^{(\mathrm{k})}(2)\chi^{(\mathrm{k})}(1)\mathrm{i}} \quad \cdots \quad \mathrm{w}^{(\mathrm{k})}_{\psi^{(\mathrm{k})}(2)\chi^{(\mathrm{k})}\left(\mathrm{L}_{\mathrm{x}}^{(\mathrm{k})}\right)\mathrm{i}} \quad \cdots \quad \cdots \quad \mathrm{w}^{(\mathrm{k})}_{\psi^{(\mathrm{k})}\left(\mathrm{L}_{\mathrm{y}}^{(\mathrm{k})}\right)\chi^{(\mathrm{k})}\left(\mathrm{L}_{\mathrm{x}}^{(\mathrm{k})}\right)\mathrm{i}}\right]^{\mathrm{T}}.$$

[0119] The subband adaptive filters $W^{(k)}$ at a time step n+1 may be updated using a known method, such as an LMS algorithm as shown in fig. 4, according to

$$\mathrm{W}^{(\mathrm{k})}(\mathrm{n}+1) = \mathrm{W}^{(\mathrm{k})}(\mathrm{n}) - \mu^{(\mathrm{k})}\mathrm{R}^{(\mathrm{k})\mathrm{H}}(\mathrm{n})\mathrm{e}^{(\mathrm{k})}(\mathrm{n}),$$

wherein $\mu^{(k)}$ is a step size, also called as a convergence gain or a learning rate.

[0120] When a proper subset of the subband reference signals are selected and used for updating the subband adaptive filters for a subband, as in fig. 4, the subband reference matrix $R^{(k)}$ may have a reduced size comparing to the example of fig. 3.

[0121] Alternatively or in combination, when a proper subset of the subband error signals are selected and used for updating the subband adaptive filters for a subband, as in fig. 4, the subband reference matrix $R^{(k)}$ may have a reduced size comparing to the example of fig. 3.

[0122] Alternatively or in combination, when a proper subset of the subband adaptive filters are updated on a subband, the subband reference matrix $R^{(k)}$ may have a reduced size comparing to the example of fig. 3.

[0123] Consequently, the above formulations related to the subband reference matrix $R^{(k)}$ also differ from that of the example of fig. 3.

[0124] Further, according to the example of fig. 4, only a selected subset of the subband adaptive filters may be updated for a subband.

[0125] Once the selected subband adaptive filters $W^{(k)}$ are updated, the fullband adaptive filters W may be reconstructed by a known weight or frequency stacking scheme 7, as shown in fig. 4. Since only some of the subband adaptive filters may be updated on at least one subband, the reconstruction scheme 7 is performed by only using the updated subband adaptive filters. There is no need to use the non-updated subband adaptive filters for reconstructing the fullband adaptive filters, because their coefficients are useless, e.g. being zeros.

[0126] Fig. 5 is another example of a noise controlling system.

[0127] The example of fig. 5 differs from the examples of figs 3-4 in that the reference signals x(n) in fig. 5 are not filtered by the secondary path model S 11 before filtering by the filter bank 10. Rather, the reference signals x(n) in fig. 5 are filtered and decimated by the filter bank 10 first. This results in L_x subband reference signals $x^{(k)}$ for each subband k.

[0128] For each subband k, a subset of the subband reference signals may be selected, e.g., by the reference signal selecting unit 5 of a control circuit (not shown). For each subband k, the selected subset of the subband reference signals may be filtered by a subband secondary path model $\hat{S}^{(k)}$ before updating the subband adaptive filters $W^{(k)}$.

[0129] That is, in the examples of figs 3-4, all the reference signals x(n) are filtered by the secondary path model S. However, in the example of fig. 5, for each subband k, only the selected subband reference signals are filtered by the subband secondary path model $\hat{S}^{(k)}$. The subband secondary path model are subband equivalent of the secondary path model S 11 of fig. 4. For example, the subband secondary path model $\hat{S}^{(k)}$ may be obtained by filtering the secondary path model S 11 of fig. 4 by the filter bank 10.

[0130] This may be advantageous as by filtering only a selected subset of subband reference signals, the computational cost may be further reduced.

[0131] Each of the subband secondary path model $\hat{S}^{(k)}$ may be modelled with J_{SAF} coefficients. Due to the decimation factor applied during subband filtering by the filter bank 10, J_{SAF} may be smaller than J. For example, J_{SAF} may be taken as a value related to J and K, such as $J_{SAF} = 4J/K$.

[0132] Further, a memory with a smaller size can be used for storing the subband secondary path model $\hat{S}^{(k)}$, comparing to a memory for storing the secondary path model S 11 in figs 3-4.

[0133] The selected subset of reference sensors may be more important than other unselected reference sensors in noise reduction in a frequency range corresponding to the subband.

[0134] Further, with the system decomposition, it is straightforward to estimate a maximum performance that the ANC system can achieve in a frequency range corresponding to one subband based on only the selected subset of the reference sensors.

[0135] According to the noise controlling systems of figs 4-5, the ANC system has L_x reference sensors. For a subband k, corresponding to a specific frequency range, a selected number $L_x^{(k)}$ reference sensors may be selected by the method of fig. 2, wherein $L_x^{(k)}$ is equal to or smaller than L_x. That is, $L_x^{(k)} \leq L_x$.

[0136] The subset of reference sensors may be defined by the function $\chi^{(k)}$. The function $\chi^{(k)}$ may be defined by performing the method of fig. 2, wherein the reference signals corresponding to the reference sensors k may be considered as the plurality of inputs of the MISO system of fig. 2. The frequency range in which the coherence should be maximized may correspond to the frequency range of a subband of interest in the examples of figs 3-5.

[0137] The function $\chi^{(k)}$ can then be defined as a function that maps $\{1,2,\ldots,L_x^{(k)}\}$ to the first $L_x^{(k)}$ ordered input indexes. For example, the inputs, i.e. the reference signals, are ordered as X_2, X_3, X_5, X_1, X_4 after they are selected and sorted by the method of fig. 2. If it is desired to select a top three inputs which contribute most ($L_x = 3$), the function χ^k would be defined as $\chi^k(1) = 2$, $\chi^k(2) = 3$, $\chi^k(3) = 5$, corresponding to the ordered inputs.

[0138] The performance that the ANC system can achieve for the subband k can be directly evaluated. For simplicity, the following indexes are chosen to be as a final order of all the inputs after they are sorted, as the example in fig. 2. For simplicity, it is considered that in the example of fig. 5, there are L_x subband reference signals per subband.

[0139] The first $L_x^{(k)}$ subband reference signals of the sorted inputs may be selected, which may contribute to the following part of the output auto spectrum S_{yy} according to

$$S_{yy,L_x^{(k)}\text{inputs}} = \sum_{i=1\ldots}|L_{iy}|^2 S_{ii.(i\text{-}1)!}\ .$$

[0140] The partial coherences can be determined based on the conditioned spectra according to

$$\gamma^2_{iy.(i-1)} = \frac{|S_{iy.(i-1)!}|^2}{S_{ii.(i-1)!}S_{yy.(i-1)!}}.$$

[0141] The multiple coherence from the $L_x^{(k)}$ selected (subband) reference signals to the output can be determined based on the partial coherences according to

$$\gamma^2_{y:L_x^{(k)}inputs} = 1\text{-}\prod\nolimits_{i=1\ldots}(1\text{-}\gamma^2_{iy.(i-1)!}).$$

[0142] Thus, a maximal sound reduction that can be achieved by using the selected $L_x^{(k)}$ (subband) reference signals of the subband k can be determined according to

$$\Delta dB_{max} = 10\log\left(1-\gamma^2_{y:L_x^{(k)}inputs}\right).$$

$L_x^{(k)}$ may be determined as the number of (subband) reference signals needed to achieve a certain level of noise reduction for a subband.

[0143] Selecting a subset of (subband) reference signals may be performed for each subband in order to determine a subset of subband reference signals for each subband. Preferably, the selection may be performed for only the first $\frac{K}{2}+1$ subbands.

[0144] Further, based on the determined subset of subband reference signals, a subset of reference sensors corresponding to the determined subset of subband reference signals may be determined, based on a one-to-one relationship between the reference sensors and the (subband) reference signals of each subband.

[0145] Then the function $\chi^{(k)}$, $k = 1,2,\ldots,\frac{K}{2}+1,$ may be defined. Here, out of the K subbands, only the first K/2 + 1 subbands are used.

[0146] The MISO system used for illustration is a part of the ANC/ARNC system. However, the method applies analogously to other types of MISO systems.

[0147] Figs 6-7 respectively illustrate four diagrams of noise reduction measurements by selecting a subset of reference sensors in an ANC system for reducing noises within a car cockpit. There are 18 reference sensors used in the ANC system. That is, there are 18 fullband reference signals, being generated by the 18 reference sensors, respectively. Thus, taking the system of fig. 5 as an example, for each subband corresponding to a specific frequency range, there are 18 subband reference signals, each corresponding to one of the 18 fullband reference signals and/or the 18 reference sensors.

[0148] The car was moving at a speed of 40 km/h when the measurements were performed. A sound was detected at a monitor microphone 4 being closed to a driver's left ear.

[0149] In the diagrams of figs 6-7, the subband reference signals are listed by a decreased contribution to the sound detected at the monitor 4. That is, the first reference signal "50:RightFrontWheel_Body_x:+X" in fig. 6 and the first reference signal "34:RightRearWheel_wisebone_z:+Z" in fig. 7 contribute most to the detected sound, respectively. The reference signals are named after the position and direction of the corresponding reference sensors generating the reference signals. The reference sensors were accelerometers placed around the different wheels on the body of the car, the wishbones, or the dampers, in a forward (+X), a lateral (+Y) or an upward (+Z) direction. Here, the forward direction is the forward moving direction of the car.

[0150] In fig. 6, the analysis is performed on a first subband corresponding to a frequency range of 160-174 Hz. In fig. 7, the analysis is performed on a second subband corresponding to a frequency range of 220-234 Hz.

[0151] The acoustic spectra of figs 6-7 are reconstructed by selecting various numbers of subband reference signals, optimized over the respective frequency ranges, for the sound detected at monitor 4.

[0152] From figs 6-7, it is clear that the top four subband reference signals which contribute the most to the sound detected at the monitor position 4, in the first subband, i.e. 160-174 Hz, are different from those in the second subband, i.e. 220-234 Hz. In the first subband, the four most significant subband reference signals consist of reference signals generated by the reference sensors oriented in the forward (+X) and lateral (+Y) directions. While in the second subband, the four most significant subband reference signals consist of reference signals generated by the reference sensors oriented in an upwards direction (+Z).

[0153] The differences do reflect an excitation of different dominating modes. For example, the dominating modes in the frequency range of 160-174 Hz are front-back and lateral modes, and the dominating modes in the frequency range of 220-234 Hz are vertical modes.

[0154] Thus, for the first and second subbands corresponding to these two frequency ranges, in order to optimise the system for reducing noises at the monitor microphone 4, it is possible to use a subset, e.g., the top four or top eight subband reference signals, of the total 18 subband reference signals, indicated in figs 6-7.

[0155] There is normally more than one monitor microphone within the acoustic cavity, e.g., the car cockpit. The method may be performed for different monitor microphones provided at different positions.

[0156] The method may be performed to span a larger frequency range, i.e. for a plurality of subbands of interest to select a subset of the reference signals for each of these subbands.

[0157] The reference signals and/or the subband reference signals may be ordered according to a value, e.g., an average sound level representing the sounds detected at more than one monitor positions.

[0158] Figs 8-9 illustrate diagrams of measured SPL values.

[0159] A sound pressure or acoustic pressure is a local pressure deviation from the ambient, average or equilibrium, atmospheric pressure, caused by a sound wave. In the air, the sound pressure can be measured using e.g., a microphone. The sound pressure level, SPL, or the acoustic pressure level, is a logarithmic measure of the effective pressure of a sound relative to a reference value.

[0160] Figs 8-9 illustrate different SPL values without and with an active noise control with a FXLMS algorithm for a road noise within a car. The algorithm is configured to control only the noise for the subbands around a resonance frequency at 160 Hz, and a resonance frequency at 230 Hz, respectively.

[0161] The car is provided with 8 reference sensors placed around the different wheels on the body of the car, the wishbones, or the dampers, wherein 4 in a forward (X) and lateral (Y) directions, and 4 in an upward direction (Z). Here, the forward direction is the forward moving direction of the car.

[0162] Among the 4 reference sensors in the forward (X) and lateral (Y) directions, 2 reference sensors are in the forward (X) direction and 2 reference sensors are in the lateral (Y) direction.

[0163] The car is also provided with 4 sound sources for generating secondary noises, e.g., loudspeakers, and 6 error sensors, e.g., control microphones.

[0164] The results shown in figs 8-9 are for the error sensor number 3, being placed on a roof over a driver's head.

[0165] The diagrams of figs 8a-8c illustrate SPL values of without and with the active noise control by using all 8 reference sensors, by using only the 4 reference sensors in the forward and lateral directions, and by using only the 4 reference sensors in the upward direction, respectively.

[0166] The diagrams of figs 9a-9c illustrate SPL values of without and with the active noise control by using all 8 reference sensors, by using only the 4 reference sensors in the forward and lateral directions, and by using only the 4 reference sensors in the upward direction, respectively.

[0167] In figs 8a-8c, the algorithm is configured to control only the subbands corresponding to a frequency around the resonance at 160 Hz, while in figs 9a-9c the algorithm is configured to control only the subbands corresponding to a frequency around the resonance at 230 Hz.

[0168] From fig. 6, it is known that at around 160 Hz, the resonance can be best represented by the reference signals in the forward and lateral directions. In connection with the measured SPL values in figs 8a and 8b, it is clear that using only these 4 reference sensors in the forward and lateral directions, the noise cancellation resulted is almost as good as using all 8 reference sensors.

[0169] Similarly, from fig. 7, it is known that at around 230 Hz, the resonance can be best represented by the reference signals in the upward direction. In connection with the measured SPL values in figs 9a and 9c, it is clear that using only these 4 reference sensors in the upward direction, the noise cancellation resulted is almost equal to a maximal noise reduction by using all 8 reference sensors.

[0170] Thus, at least for these subbands, there is no need to involve all the subband reference signals to achieve a good noise reduction. Rather, by selecting a subset of subband reference signals, a good noise reduction may be achieved. Meanwhile, the computational cost involved may be reduced and the convergence speed may be improved.

[0171] The subset, preferably a proper subset, of error sensors, and a subset, preferably a proper subset, of sound sources for a subband k, may be determined by different methods, e.g., by an optimal spatial matching of a primary sound field by a secondary sound field, for a specific frequency range of the subband k.

[0172] The sound fields within an acoustic cavity, e.g., a car cockpit, may be mainly governed by resonant acoustic modes, which are dependent on the frequency of the acoustic waves. Thus, for different subbands corresponding to different frequency ranges, the sound fields may be governed by different acoustic modes.

[0173] The positions of the error sensors and/or sound sources may be selected to match the acoustic modes over a whole frequency range of the noise to be cancelled and different operating conditions, such as a moving speed of a car.

[0174] However, for the subband k, it is common that not all provided error sensors and/or sound sources may be needed, as the acoustic modes of the frequency range of the subband k may be different from the acoustic modes over the whole frequency range of the noise to be cancelled, especially at lower frequencies, where the acoustic modes are less complex, e.g., 20 to 100 Hz.

[0175] Thus, for the subband k, it is possible to use only the subband reference signals decomposed from the reference signals generated by those reference sensors which are needed for the subband k. That is, the subband reference signals decomposed from the reference signals generated by those reference sensors which are not needed, can be discarded, when processing with the subband reference signals, e.g. when updating the subband adaptive filters.

[0176] It may be advantageous to discard those subband reference signals when they are not needed, to further reduce a size of the system for each subband. Analogously, for the subband k, it is possible to discard some subband error signals. Analogously, it is also possible not to update at least some of the subband adaptive filters corresponding to the sound sources that are not needed.

[0177] With a reduced number of subband signals, the amount of computational operations may be reduced, and the convergence speed of the method may be improved.

[0178] Fig. 10 illustrates a diagram of measured SPL values when different ANC methods is performed or none ANC method is performed, i.e. ANC off.

[0179] The SPL values in fig. 10 are measured at a position close to a front passenger's ear within a small electric car, which is provided with 8 reference sensors placed close to the wheels, 4 loudspeakers and 6 microphones. The car was moving forward at a speed of 40 km/h when measurements were performed.

[0180] The SPL values, without any active noise control, with an active noise control method using a fullband FXLMS algorithm, with an active noise control method using a subband FXLMS algorithm with all subband reference signals used, and with an active noise control method using a subband FXLMS algorithm with a selected subset of subband reference signals for selected subbands, are presented in fig. 10. Here, the selected subset of subband reference signals is about a half of all the subband reference signals for the selected subbands.

[0181] From fig. 10, it is clear that a similar level of noise reduction can be achieved by the active noise control method using a subband FXLMS algorithm using the selected subset of subband reference signals for selected subbands, comparing with other two active noise control methods. However, using a smaller and optimized subset of subband reference signals may result in a much lower computational cost.

[0182] In this example, only the subset of the reference sensors are selected. That is, all the subband error signals are used and all the subband adaptive filters are updated for all sound sources. However, a subset of error sensors and a subset of sound sources for the subband k may be selected to further reduce the computational cost.

[0183] The step size of each individual subband may be adjusted based on the subband reference signals to reduce the spectral range or eigenvalue spread of the filtered reference matrix in each subband, to improve the convergence of the subband FXLMS algorithm. However, the step size is not adjusted in this example, explaining the similar performances between the fullband FXLMS and the subband FXLMS.

[0184] Fig. 11 visualises an example the function $\chi^{(k)}$ for selecting the subset of the reference sensors to be active on each subband for the example shown in fig. 10.

[0185] The y-axis of fig. 11 is a reference sensor index. That is, each one of the numbers 1 to 8 refers to one of the eight reference sensors of fig. 10. The x-axis of fig. 11 is a subband index. An algorithm with 128 subbands was used in this example, where all information may be considered to be contained within the first 65 subbands.

[0186] The function $\chi^{(k)}$ for selecting a subset of the reference sensors according to fig. 10 is defined as fig. 11.

[0187] For example, in subband 1, the subband reference signals derived from the reference sensors 1, 2, 3, 4, 7 and 8 are selected, while the subband reference signals derived from the reference sensors 5 and 6 are not selected.

[0188] For example, in subbands 20 and 21, none of the reference sensors is selected. So do the subbands 24 to 65. The subband adaptive filters on these subbands having no subband reference signals due to no reference sensors being selected may not be updated.

[0189] The computational costs for different ANC methods are listed and compared in below Form. 1.

[0190] The example ANC system used has M sound sources, L_x reference sensors, L_e error sensors, K subbands, I taps for fullband adaptive filters, and J taps for secondary path models. The decimation rate D is taken as K/4. The number of taps for subband adaptive filters and the secondary path models are taken respectively as ISAF=4I/K and JSAF=4J/K. The numbers of multiplications per sample required in each step of the proposed method and other known ANC methods are listed in Form. 1.

[0191] It is assumed that in the proposed method, the signals corresponding to about 50% of the reference sensors, error sensors and sound sources are selected for each subband, and about only 50% of all the subbands are updated.

Form. 1 Computational cost

	The ANC method with a fullband FXLMS	The ANC method with a subband FXLMS with fullband secondary path modelling	The ANC method with a subband FXLMS with subband secondary path modelling	Proposed ANC method with a subband FXLMS with subband signal selector (fig. 4)
Reference signals filtering	$MLxLeJ$	Same	$\frac{3ML_eL_xJSAF\left(\frac{K}{2}+1\right)}{D}$	$\frac{3\frac{M}{2}\frac{L_e}{2}\frac{L_x}{2}JSAF\left(\frac{K}{4}+1\right)}{D}$
Subband analysis	-	$(L_e + ML_eL_x)4log2(K)$	$(L_e + L_x)4log2(K)$	$(L_e + L_x)4log2(K)$
Adaptive filter update	$ML_xI(1 + L_e)$	$(3L_eML_xISAF + 2ML_xISAF)\frac{\left(\frac{K}{2}+1\right)}{D}$	$(3L_e\,ML_xISAF + 2ML_xISAF)\frac{\left(\frac{K}{2}+1\right)}{D}$	$\left(3\frac{Le}{2}\frac{M}{2}\frac{Lx}{2}\,ISAF + 2\frac{M}{2}\frac{Lx}{2}ISAF\right)\frac{\left(\frac{K}{4}+1\right)}{D}$
Weight stacking (subband synthesis)	-	$\frac{ML_x}{D}\left(4ISAF\left(\frac{K}{2}+1\right)log2(2ISAF) + 2Ilog2(2I)\right)$	$\frac{ML_x}{D}\left(4ISAF\left(\frac{K}{2}+1\right)log2(2ISAF) + 2Ilog2(2I)\right)$	$\frac{M/2\,L_x/2}{D}\left(4ISAF\left(\frac{K}{2}+1\right)log2(2ISAF) + 2Ilog2(2I)\right)$
Fullband control signal generation	ML_xI	ML_xI	ML_xI	ML_xI
Total computational cost for $M = 4$, $L_x = 8$, $L_e = 6$, $J = 256$, $I = 256$, $K = 128$	114688	86216	41272	12112

[0192] The fullband secondary path modelling means that the reference signals x(n) may be first filtered by the secondary path model S 11. The filtered reference signals x'(n) may then be filtered by the filter bank 10 consisting of K subbands. Then, for each subband *k,* the subset of the subband reference signals of the subband *k* may be selected.

[0193] The subband secondary path modelling means that the reference signals x(n) are not filtered by the secondary path model $\hat{S}$ 11 before filtering by the filter bank 10. Rather, the reference signals x(n) are filtered and decimated by the filter bank 10 first. For each subband k, a subset of the subband reference signals may be selected. The selected subset of the subband reference signals may be filtered by the subband secondary path model $\hat{S}^{(k)}$ before updating the subband adaptive filters $W^{(k)}$, as in fig. 5.

[0194] Fig. 12 is a diagram of numbers of multiplications per sample needed for the different methods of Form. 1, wherein M = 4, L_x = 8, L_e = 6, J = 256 and I = 256.

[0195] The dotted line represents the number of multiplications per sample of the ANC method with standard fullband FXLMS.

[0196] The upper solid line represents the number of multiplications per sample of the ANC method with subband FXLMS using fullband secondary path modelling.

[0197] The upper dashed line represents the number of multiplications per sample of the ANC method with subband FXLMS using subband secondary path modelling.

[0198] The lower solid line represents the number of multiplications per sample of the proposed ANC method with subband FXLMS using fullband secondary path modelling and subband signal selection, as in fig. 4.

[0199] The lower dashed line represents the number of multiplications per sample of the proposed ANC method with subband FXLMS using subband secondary path modelling and subband signal selection, as in fig. 5.

[0200] When there are 128 subbands, the proposed ANC method with subband FXLMS using subband secondary path modelling and subband signal selection represents a reduction in computational cost of a factor of 3.4 compared to the same algorithm without subband signal selection, and a factor 9.5 compared to the ANC method with standard fullband FXLMS algorithm.

[0201] For the example shown in fig. 10, the computational cost may be reduced by a factor of 6 compared to the ANC method with fullband algorithm and a factor of 2.5 compared to ANC method with the standard subband algorithm.

[0202] Additional computational cost reduction may be achieved if a subset of the error sensors and/or a subset of the sound sources to be active on each subband are selected for updating the subband adaptive filters.

[0203] Thus, the method and apparatus for selecting a subset of a plurality of inputs may be advantageous in characterising a MISO system. The method and apparatus for selecting a subset of a plurality of inputs may be applied in different types of MISO systems.

Claims

1. A method for selecting a subset of a plurality of inputs of a Multiple-Input-Single-Output, MISO, system, the MISO system comprising:

the plurality of inputs X_i, wherein i = 1, 2,..., n, and
a single output Y;
the method comprising steps in a following order:

1) calculating a coherence value representing a coherence level between each of the plurality of inputs X_i and the single output Y;
2) among the plurality of inputs X_i, selecting an input having a largest coherence value;
3) creating a remaining group of inputs, wherein the remaining group of inputs consists all inputs of the plurality of inputs X_i except the previously selected input(s);
4) for each input of the remaining group of inputs,
generating a corresponding conditioned input by conditioning the input;
5) for each conditioned input,
calculating a partial coherence value representing a coherence level between the conditioned input and the single output Y;
6) among the remaining group of inputs, selecting an input corresponding to a conditioned input having a largest partial coherence value.

2. The method as claimed in claim 1, further comprising
repeating the steps 3)- 6), until the remaining group of inputs consisting of a last one input, and selecting the last one input.

3. The method as claimed in claim 2, further comprising
prior to selecting the last one input, performing steps 4) -5) for conditioning the last one input and calculating a partial coherence value between the conditioned last one input and the single output Y.

4. The method as claimed in claim 1, further comprising
repeating the steps 3)- 6), until the largest partial coherence value calculated at step 5) being smaller than a threshold.

5. The method as claimed in any one of claims 1-4, further comprising
sorting the plurality of inputs X_i based on a sequence that each input of the plurality of inputs X_i is selected, such that the input selected at the step 2) has a highest ranking.

6. The method as claimed in any one of claims 1-5, wherein a relationship between the conditioned inputs and the single output Y is

$$Y = \Sigma_{i=1}^{n} L_{iy} X_{i.(i-1)!} + N,$$

wherein $X_{i\cdot(i-1)!}$ refers to an input X_i conditioned by the previously selected input(s) $X_{(i-1)!}$,
L_{iy} refers to a transfer function, and
N is a constant.

7. The method as claimed in claim 6,
wherein the step 4) of generating the conditioned inputs $X_{i\cdot(i-1)!}$ comprises
conditioning each input of the remaining group of inputs by a previously selected input according to

$$X_{i.(i-1)!} = X_{i.(i-2)!} - L_{(i-1)i}X_{(i-1)(i-2)!}.$$

8. The method as claimed in any one of claims 1-7, further comprising
prior to the step 1) of calculating the coherence value, arranging the plurality of input X_i in an arbitrary order.

9. The method as claimed in any one of claims 1-8, further comprising:
prior to the step 1) of calculating the coherence value, performing an optimal least-square identification on the plurality of inputs X_i and the single output *Y.*

10. A noise controlling method, comprising:

generating a plurality of reference signals representing a plurality of primary noises;
generating a secondary noise in response to a control signal, for cancelling the plurality of primary noises;
generating an error signal representing a superposition of the plurality of primary noises and the secondary noise at a position;
wherein the method further comprises:

generating the control signal for generating the secondary noise, by executing an adaptive subband filtering algorithm based on the plurality of reference signals and the error signal;
wherein the step of generating the control signal comprises:

decomposing the plurality of reference signals and the error signal into subband reference signals and a subband error signal, respectively, for each subband of a plurality of subbands;
updating a plurality of subband adaptive filters for at least one subband of the plurality of subbands, based on a proper subset of the subband reference signals of the at least one subband and the subband error signal of the at least one subband;
updating at least one fullband adaptive filter based on the updated plurality of subband adaptive filters;
generating the control signal by filtering the plurality of reference signals by the updated at least one fullband adaptive filter;
wherein the method further comprising:

selecting the proper subset of the subband reference signals of the at least one subband by the method as claimed in any one of claims 1-9;
wherein the subband reference signals correspond to the plurality of input X_i of the method as claimed in any one of claims 1-9; and
wherein the subband error signal corresponds to the single output *Y* of the method as claimed in any one of claims 1-9,

or,
wherein the method further comprises generating a sound signal representing a sound at a second position, and
wherein the sound signal corresponds to the single output Y of the method as claimed in any one of claims 1-9.

11. An apparatus for selecting a subset of a plurality of inputs of a Multiple-Input-Single-Output, MISO, system, the MISO system comprising:

the plurality of inputs X_i, wherein i = 1, 2, ... n, and
a single output *Y;*

the apparatus comprising a control circuit configured to perform following functions in a following order:

a coherence value calculation function configured to calculate a coherence value for each of the plurality of inputs X_i, representing a coherence level between each of the plurality of inputs X_i and the single output Y;
a coherence value selection function configured to among the plurality of inputs X_i, select an input having a largest coherence value;
a group creation function configured to create a remaining group of inputs, wherein the remaining group of inputs consists all inputs of the plurality of inputs X_i except the previously selected input(s);
a condition function configured to, for each input of the remaining group of inputs, generate a conditioned input by conditioning the input;
a partial coherence value calculation function configured to calculate a partial coherence value for each conditioned input, representing a coherence level between the conditioned input and the single output Y;
a partial coherence value selection function configured to among the remaining group of inputs, select an input corresponding to a conditioned input having a largest partial coherence value.

12. The apparatus as claimed in claim 11, wherein the control circuit is configured to perform
an iteration function configured to repeat the group creation function, the condition function, the partial coherence value calculation function, and the partial coherence value selection function,
until the remaining group of inputs consisting of a last one input, and to select the last one input,
or until the largest partial coherence value calculated by the partial coherence value calculation function being smaller than a threshold.

13. The apparatus as claimed in claim 11 or 12, wherein the control circuit is configured to perform
a sorting function configured to sort the plurality of inputs X_i based on a sequence that each input of the plurality of inputs X_i is selected, such that the input selected by performing the coherence value selection function has a highest ranking.

14. A noise controlling system, comprising:

a plurality of reference sensors configured to generate a plurality of reference signals representing a plurality of primary noises, respectively;
a sound source configured to generate a secondary noise in response to a control signal, for cancelling the plurality of primary noises;
an error sensor configured to generate an error signal representing a superposition of the plurality of primary noises and the secondary noise at a position; and
a control unit configured to generate the control signal by executing an adaptive subband filtering algorithm, based on the plurality of reference signals and the error signal;
wherein the control unit is further configured to:

decompose the plurality of reference signals and the error signal into subband reference signals and a subband error signal, respectively, for each subband of a plurality of subbands;
update a plurality of subband adaptive filters for at least one subband of the plurality of subbands, based on a proper subset of the subband reference signals of the at least one subband and the subband error signal of the at least one subband;
update at least one fullband adaptive filter based on the updated plurality of subband adaptive filters;
generate the control signal by filtering the plurality of reference signals by the updated at least one fullband adaptive filter;
wherein the noise controlling system further comprises the apparatus as claimed in any one of claims 11-13, for selecting the proper subset of the subband referent signals; and
wherein the subband reference signals correspond to the plurality of input X_i of the apparatus as claimed in any one of claims 11-13; and
wherein the subband error signal corresponds to the single output Y of the apparatus as claimed in any one of claims 11-13,

or,
wherein the noise controlling system further comprises a sensor configured to generate a sound signal representing a sound at a second position, and
wherein the sound signal corresponds to the single output Y of the apparatus as claimed in any one of claims

11-13.

15. A non-transitory computer readable recording medium having computer readable program code recorded thereon which when executed on a device having processing capability is configured to perform the method of any one of claims 1-10.

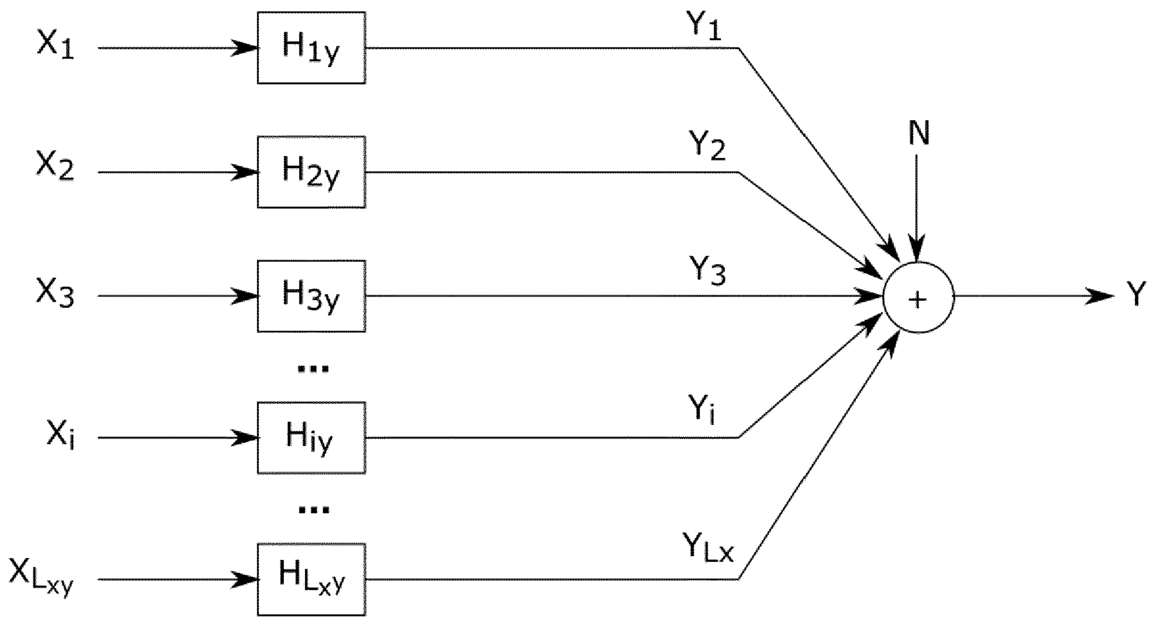

X_1
H_{1y}
Y_1
X_2
H_{2y}
Y_2
N
X_3
H_{3y}
Y_3
+
Y
...
X_i
H_{iy}
Y_i
...
$X_{L_{xy}}$
$H_{L_{xy}}$
Y_{Lx}

Fig 1a

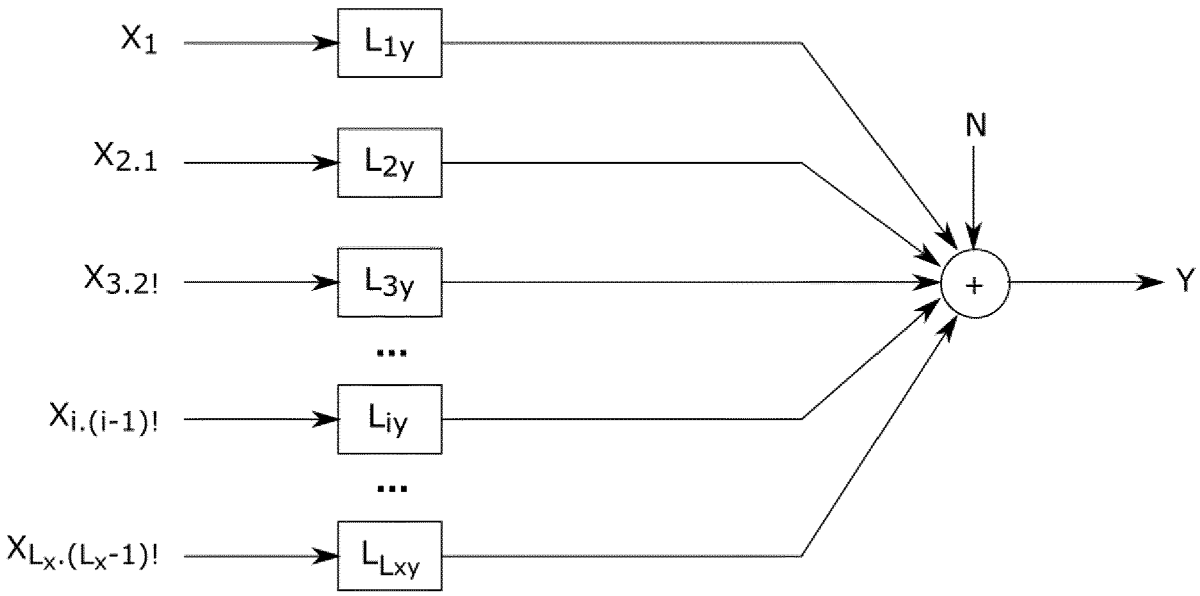

X_1
L_{1y}
$X_{2.1}$
L_{2y}
N
$X_{3.2!}$
L_{3y}
+
Y
...
$X_{i.(i-1)!}$
L_{iy}
...
$X_{L_x.(L_x-1)!}$
$L_{L_{xy}}$

Fig 1b

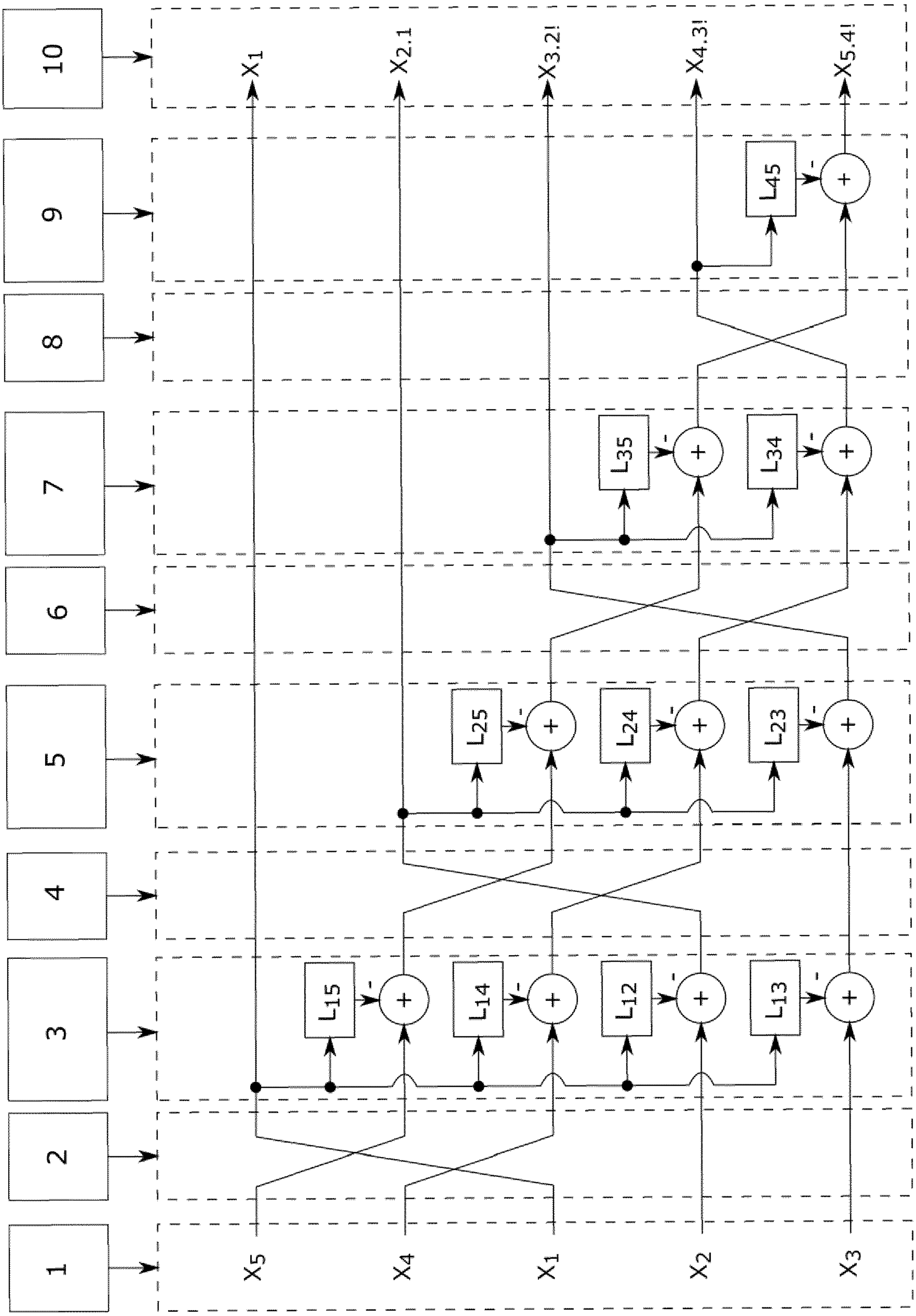

1
2
3
4
5
6
7
8
9
10
X_5
X_4
X_1
X_2
X_3
L_{15}
L_{14}
L_{12}
L_{13}
L_{25}
L_{24}
L_{23}
L_{35}
L_{34}
L_{45}
X_1
$X_{2.1}$
$X_{3.2!}$
$X_{4.3!}$
$X_{5.4!}$

Fig 2

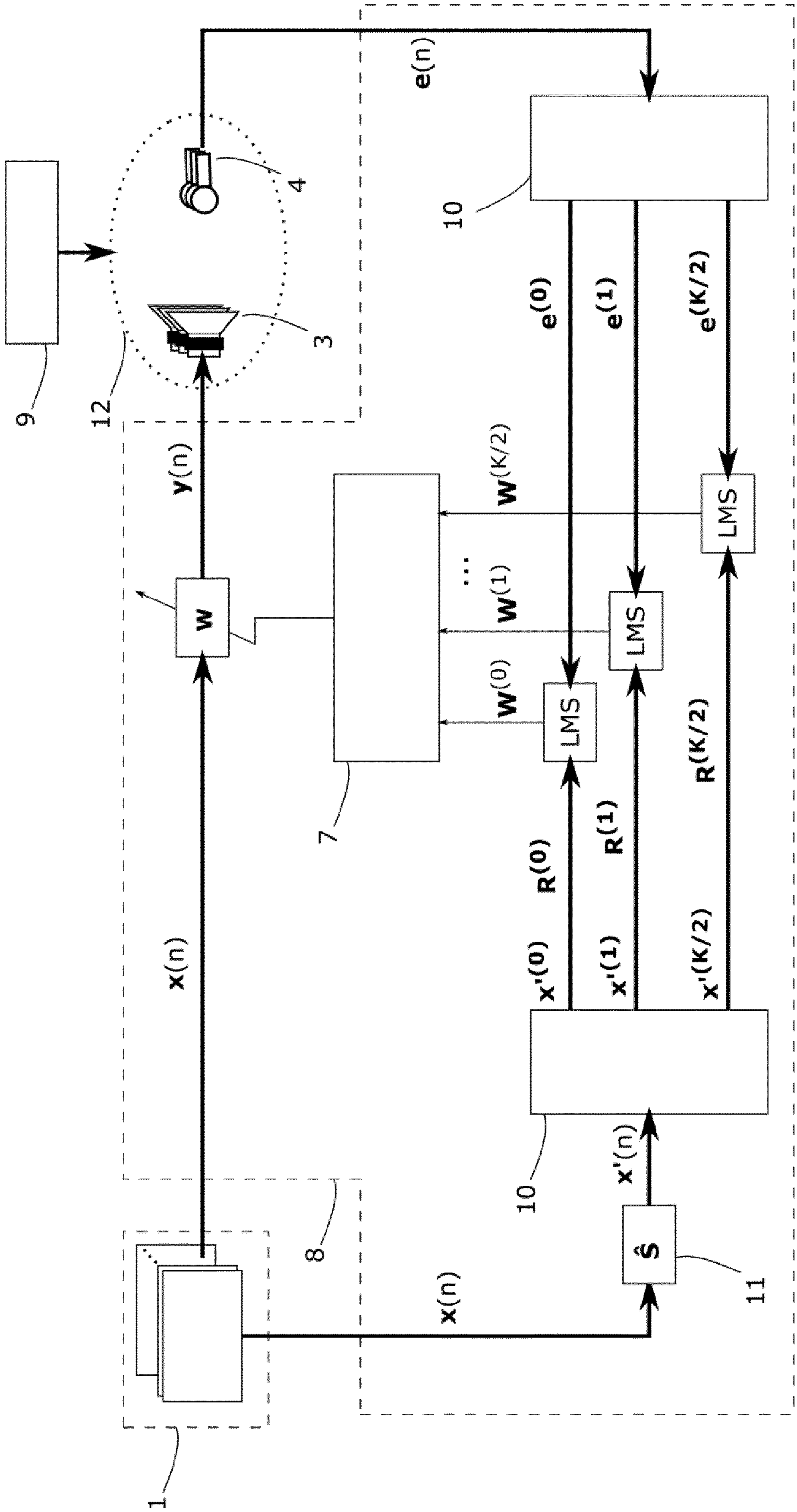
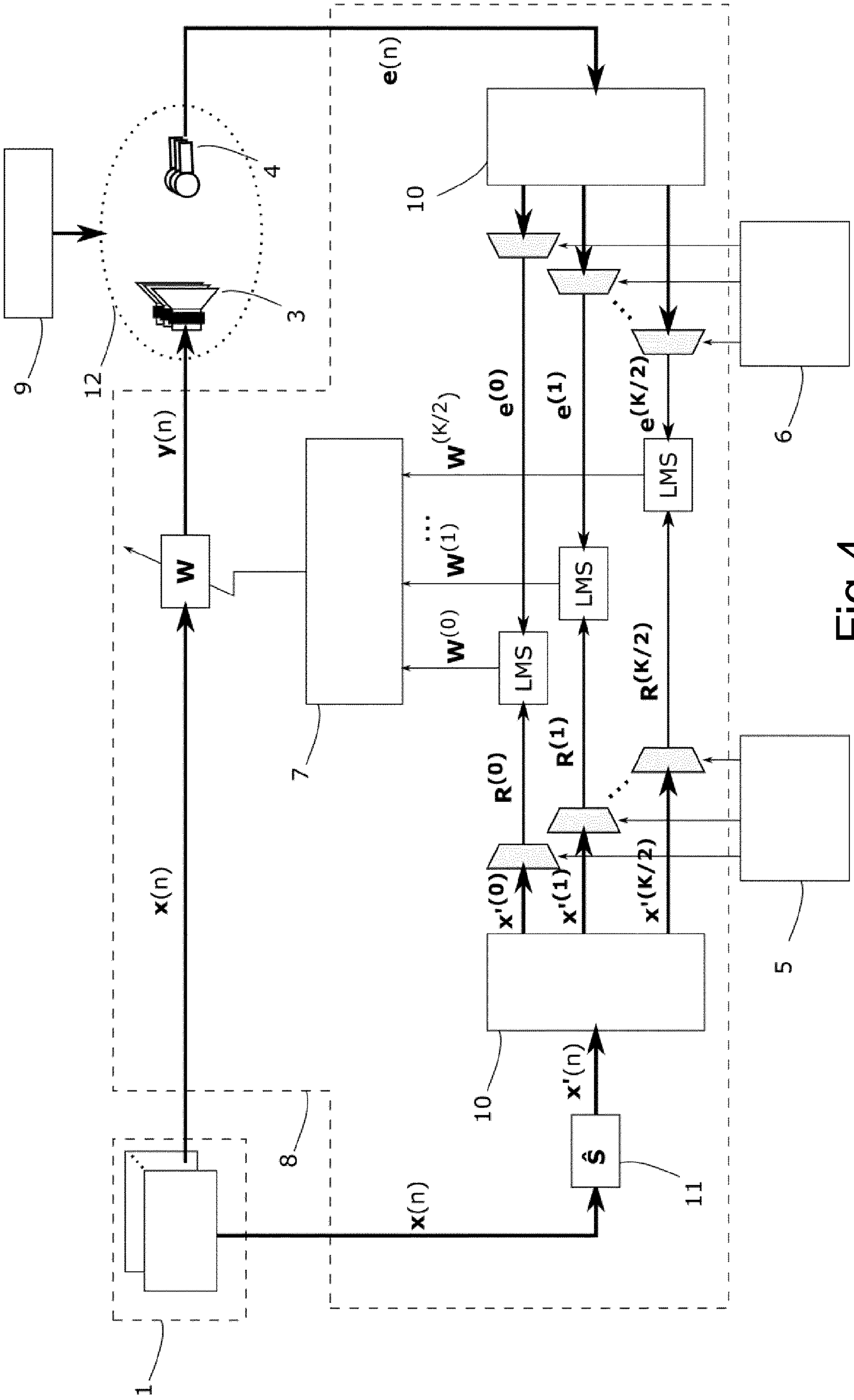

e(n)
10
4
3
9
12
y(n)
w
W(K/2)
W(1)
W(0)
...
LMS
LMS
LMS
e(0)
e(1)
e(K/2)
7
R(0)
R(1)
R(K/2)
x(n)
x'(0)
x'(1)
x'(K/2)
10
x'(n)
8
Ŝ
11
x(n)
1

Fig 3


1
8
9
12
3
4
10
10
11
5
6
7
x(n)
x(n)
y(n)
e(n)
W
Ŝ
x'(n)
x'(0)
x'(1)
x'(K/2)
R(0)
R(1)
R(K/2)
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LMS
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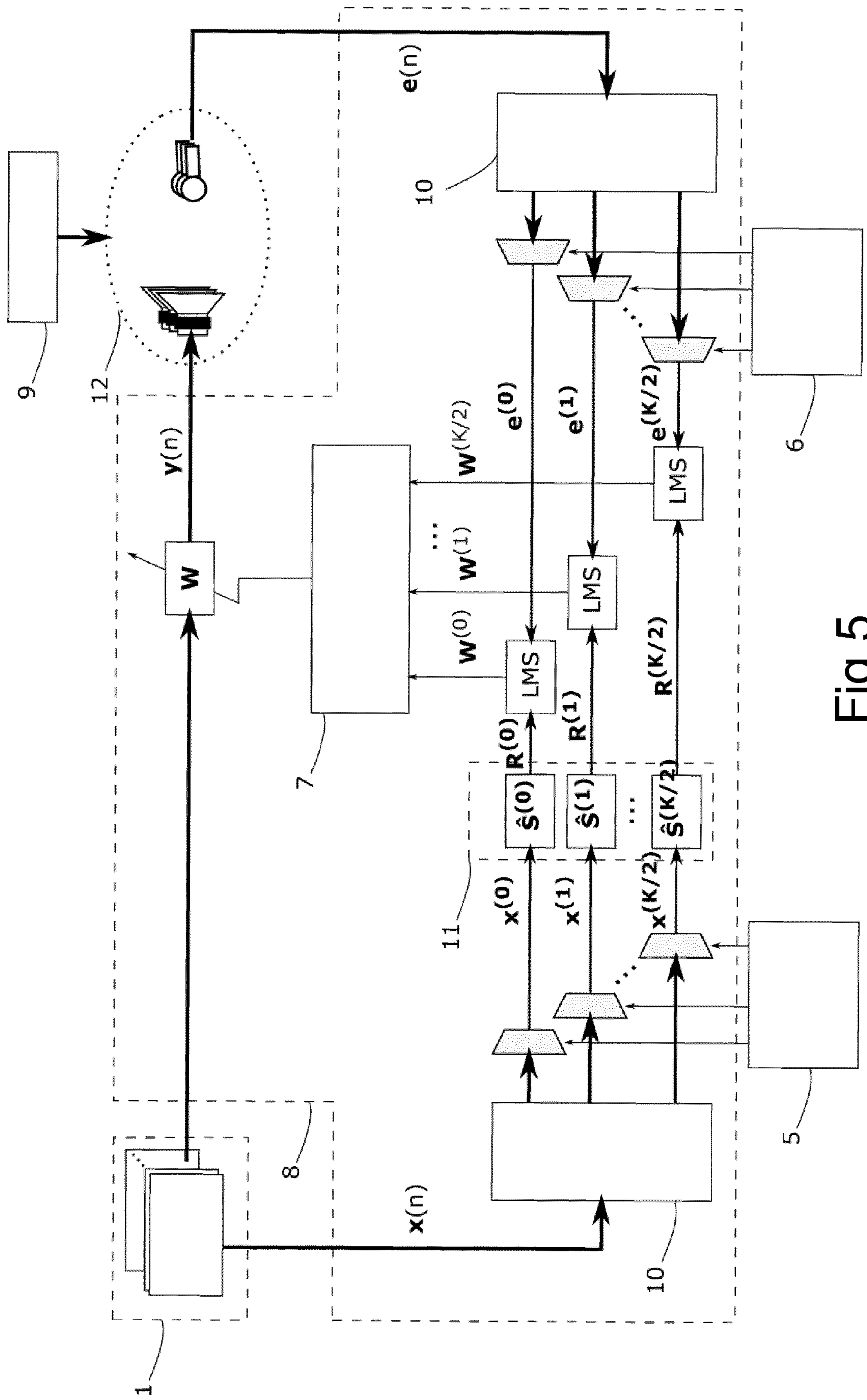
W(0)
W(1)
W(K/2)

Fig 4



e(n)
10
9
12
e(0)
e(1)
e(K/2)
y(n)
W(K/2)
W(1)
W(0)
W
LMS
LMS
LMS
6
7
R(0)
R(1)
R(K/2)
Ŝ(0)
Ŝ(1)
Ŝ(K/2)
11
x(0)
x(1)
x(K/2)
8
x(n)
10
5
1

Fig 5

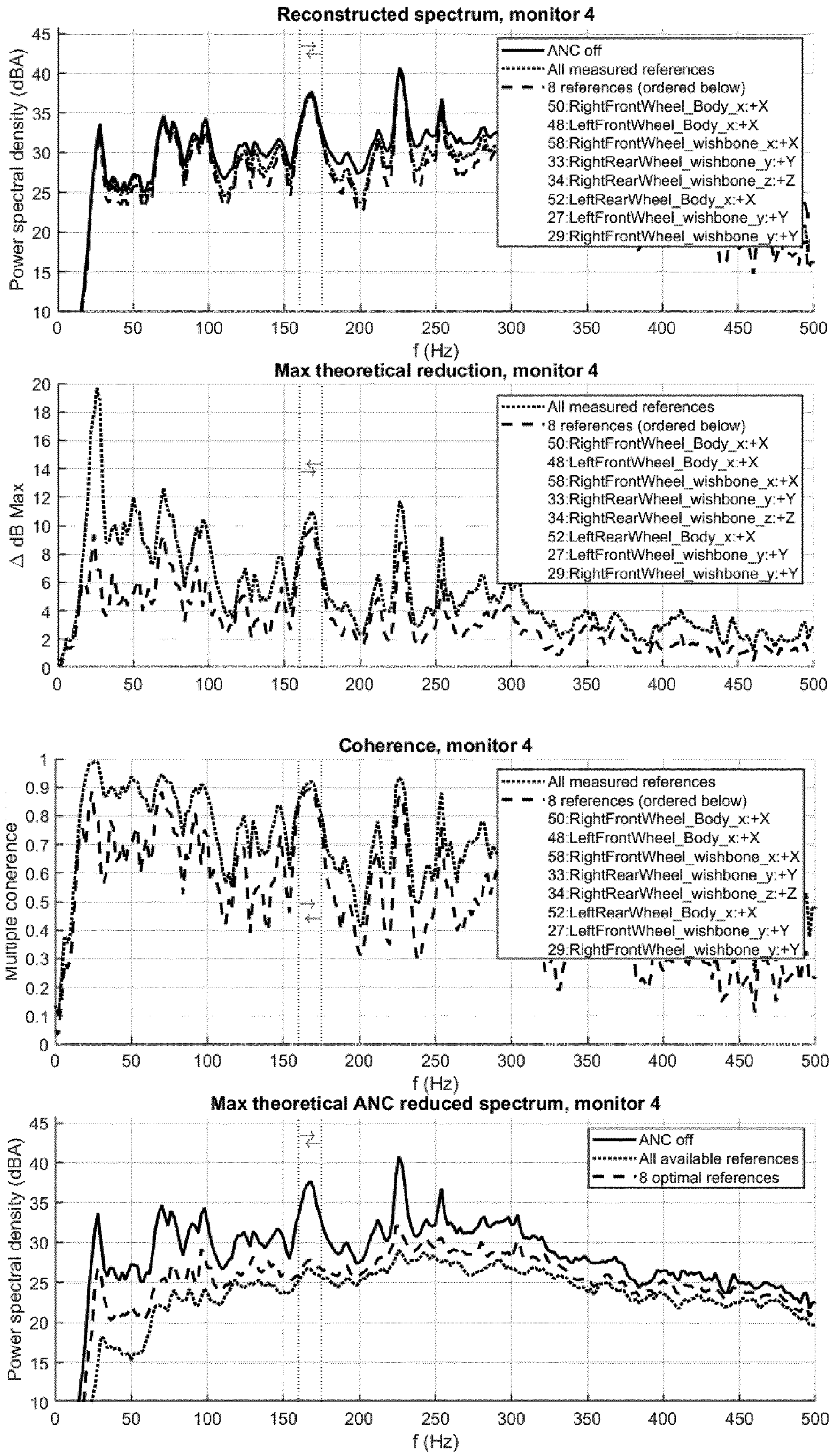

Reconstructed spectrum, monitor 4
Power spectral density (dBA)
f (Hz)
ANC off
All measured references
8 references (ordered below)
50:RightFrontWheel_Body_x:+X
48:LeftFrontWheel_Body_x:+X
58:RightFrontWheel_wishbone_x:+X
33:RightRearWheel_wishbone_y:+Y
34:RightRearWheel_wishbone_z:+Z
52:LeftRearWheel_Body_x:+X
27:LeftFrontWheel_wishbone_y:+Y
29:RightFrontWheel_wishbone_y:+Y
Max theoretical reduction, monitor 4
Δ dB Max
All measured references
8 references (ordered below)
50:RightFrontWheel_Body_x:+X
48:LeftFrontWheel_Body_x:+X
58:RightFrontWheel_wishbone_x:+X
33:RightRearWheel_wishbone_y:+Y
34:RightRearWheel_wishbone_z:+Z
52:LeftRearWheel_Body_x:+X
27:LeftFrontWheel_wishbone_y:+Y
29:RightFrontWheel_wishbone_y:+Y
Coherence, monitor 4
Multiple coherence
f (Hz)
All measured references
8 references (ordered below)
50:RightFrontWheel_Body_x:+X
48:LeftFrontWheel_Body_x:+X
58:RightFrontWheel_wishbone_x:+X
33:RightRearWheel_wishbone_y:+Y
34:RightRearWheel_wishbone_z:+Z
52:LeftRearWheel_Body_x:+X
27:LeftFrontWheel_wishbone_y:+Y
29:RightFrontWheel_wishbone_y:+Y
Max theoretical ANC reduced spectrum, monitor 4
Power spectral density (dBA)
f (Hz)
ANC off
All available references
8 optimal references

Fig 6

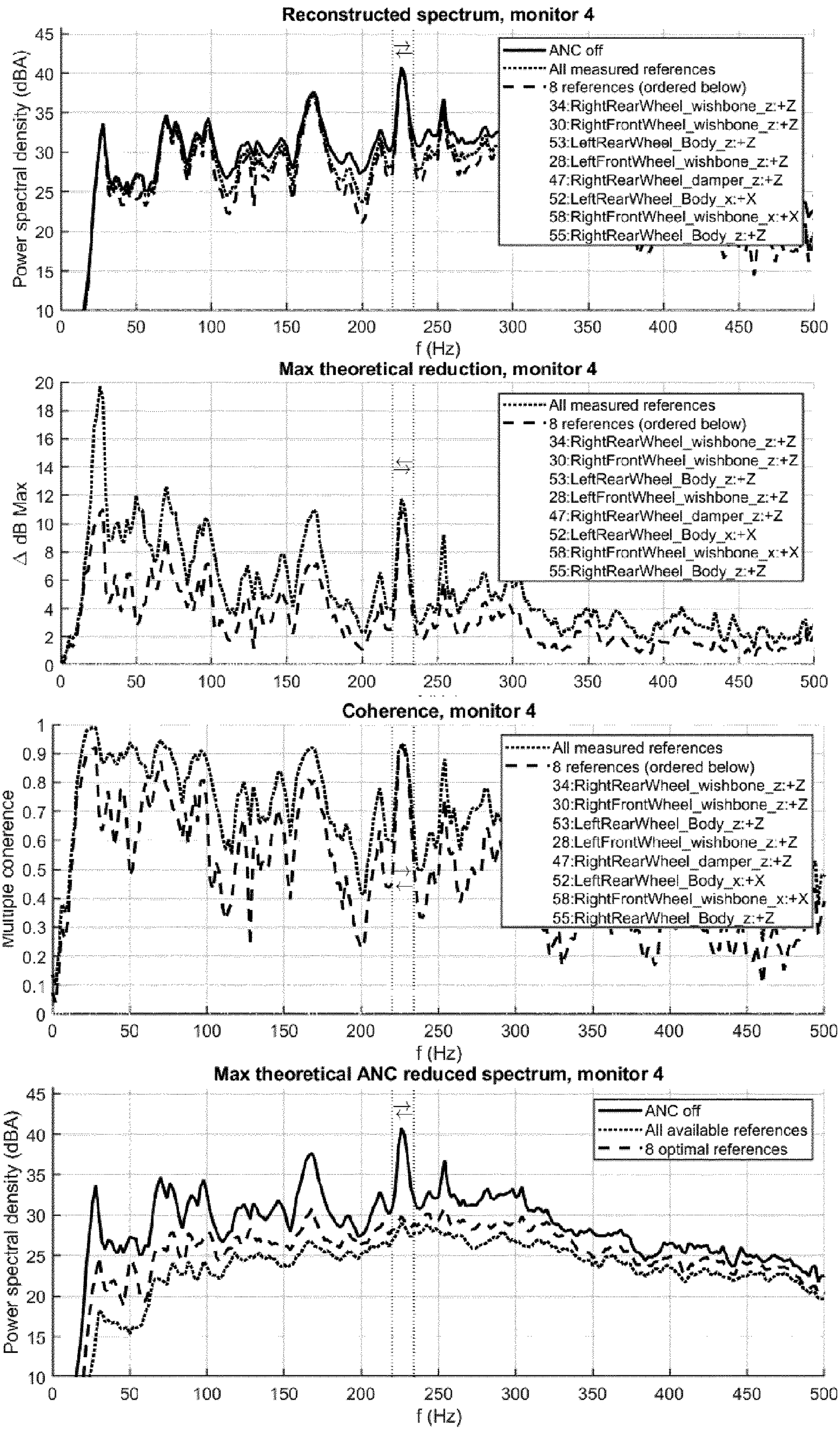

Reconstructed spectrum, monitor 4
Power spectral density (dBA)
f (Hz)
ANC off
All measured references
8 references (ordered below)
34:RightRearWheel_wishbone_z:+Z
30:RightFrontWheel_wishbone_z:+Z
53:LeftRearWheel_Body_z:+Z
28:LeftFrontWheel_wishbone_z:+Z
47:RightRearWheel_damper_z:+Z
52:LeftRearWheel_Body_x:+X
58:RightFrontWheel_wishbone_x:+X
55:RightRearWheel_Body_z:+Z
Max theoretical reduction, monitor 4
Δ dB Max
All measured references
8 references (ordered below)
34:RightRearWheel_wishbone_z:+Z
30:RightFrontWheel_wishbone_z:+Z
53:LeftRearWheel_Body_z:+Z
28:LeftFrontWheel_wishbone_z:+Z
47:RightRearWheel_damper_z:+Z
52:LeftRearWheel_Body_x:+X
58:RightFrontWheel_wishbone_x:+X
55:RightRearWheel_Body_z:+Z
Coherence, monitor 4
Multiple coherence
f (Hz)
All measured references
8 references (ordered below)
34:RightRearWheel_wishbone_z:+Z
30:RightFrontWheel_wishbone_z:+Z
53:LeftRearWheel_Body_z:+Z
28:LeftFrontWheel_wishbone_z:+Z
47:RightRearWheel_damper_z:+Z
52:LeftRearWheel_Body_x:+X
58:RightFrontWheel_wishbone_x:+X
55:RightRearWheel_Body_z:+Z
Max theoretical ANC reduced spectrum, monitor 4
Power spectral density (dBA)
f (Hz)
ANC off
All available references
8 optimal references

Fig 7

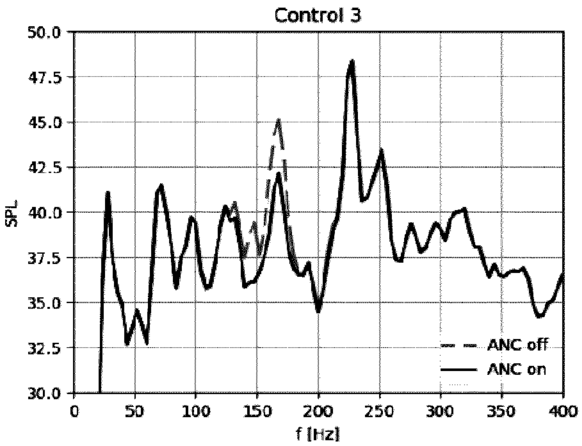


Fig 8a

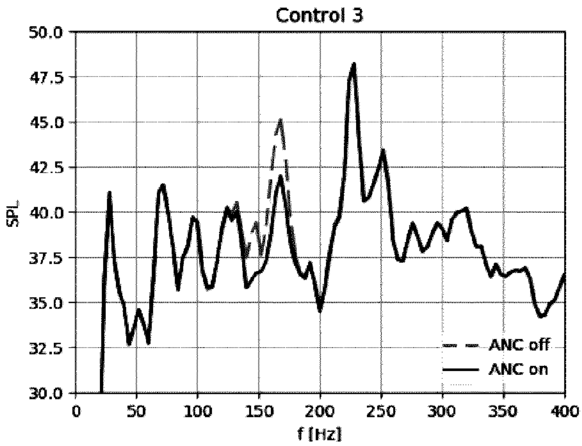


Fig 8b

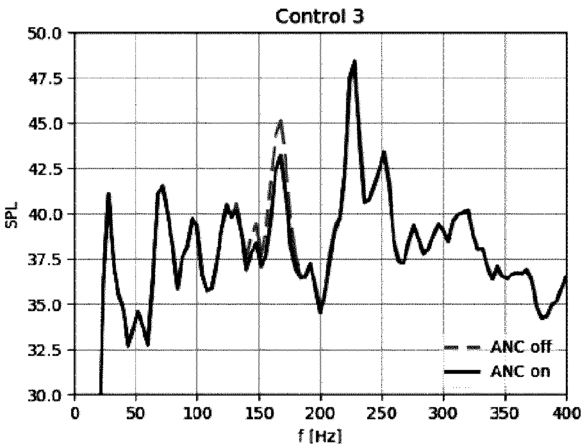


Fig 8c

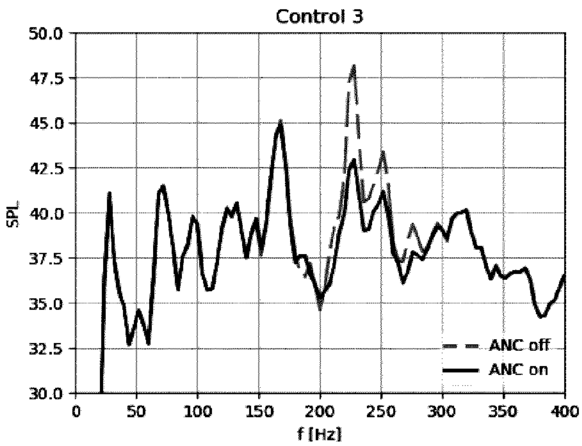


Fig 9a

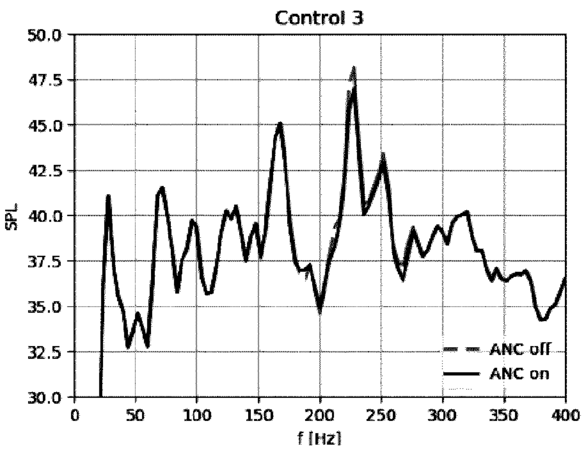


Fig 9b

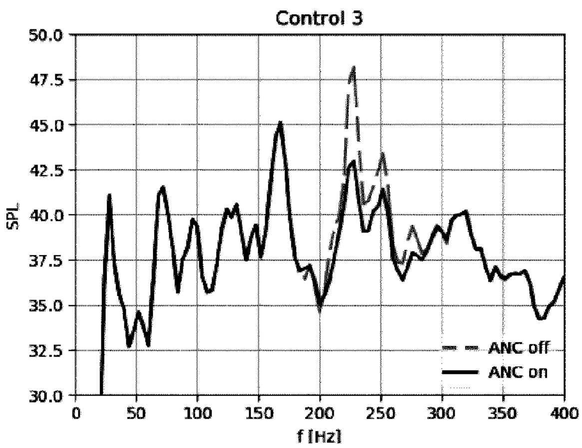
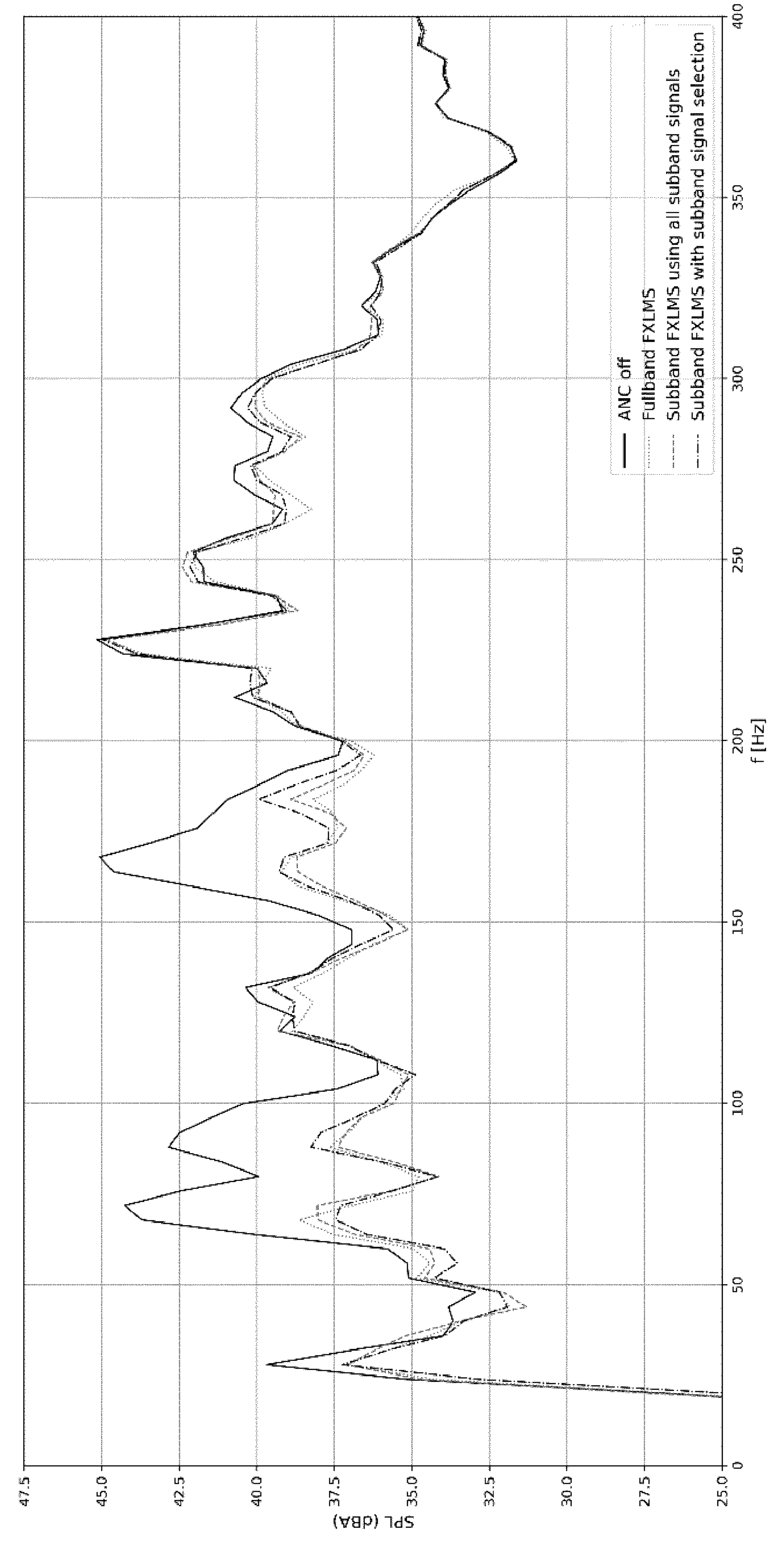


Fig 9c



ANC off
Fullband FXLMS
Subband FXLMS using all subband signals
Subband FXLMS with subband signal selection
SPL (dBA)
f [Hz]
47.5
45.0
42.5
40.0
37.5
35.0
32.5
30.0
27.5
25.0
0
50
100
150
200
250
300
350
400

Fig 10

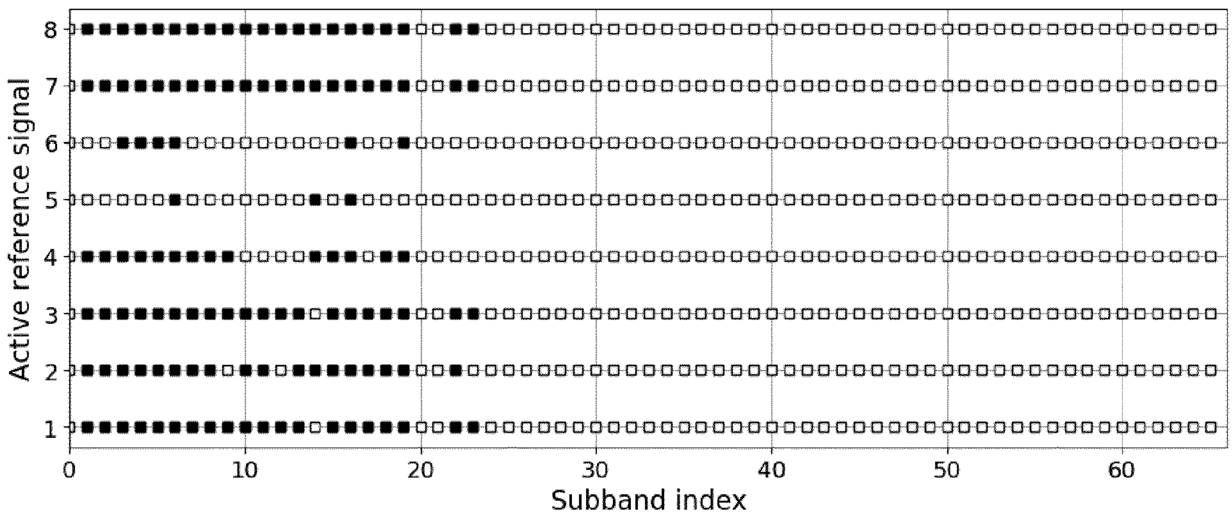

Active reference signal
Subband index
0
10
20
30
40
50
60
1
2
3
4
5
6
7
8

Fig 11

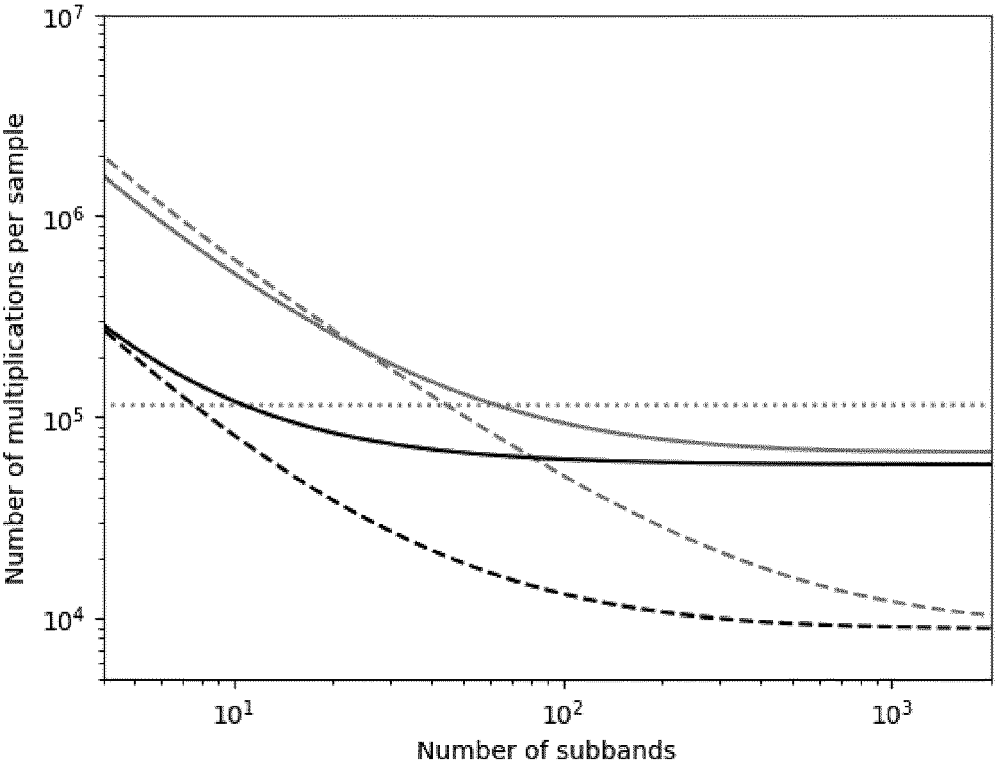

Number of multiplications per sample
Number of subbands
10^7
10^6
10^5
10^4
10^1
10^2
10^3

Fig 12

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EUROPEAN SEARCH REPORT

Application Number
EP 19 18 5733

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (IPC)
X	US 2014/086425 A1 (JENSEN THOMAS M [US] ET AL) 27 March 2014 (2014-03-27) * paragraphs [0006], [0029] - [0030] * -----	1-15	INV. G10K11/178
X	US 2019/206382 A1 (TANI MITSUHIRO [JP]) 4 July 2019 (2019-07-04) * paragraphs [0049] - [0056], [0070] - [0078]; figures 2, 6, 9 * -----	1-15	
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A	WO 2019/024984 A1 (HARMAN BECKER AUTOMOTIVE SYSTEMS GMBH [DE]) 7 February 2019 (2019-02-07) * paragraphs [0015] - [0028] * -----	1-15	
			TECHNICAL FIELDS SEARCHED (IPC)
			G10K

The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
The Hague	10 February 2020	Naujoks, Marco

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10-02-2020

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