(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) **EP 1 583 075 A1**

(12) **EUROPEAN PATENT APPLICATION**

(43) Date of publication:
**05.10.2005 Bulletin 2005/40**

(51) Int Cl.⁷: **G10K 11/178**

(21) Application number: **04076031.6**

(22) Date of filing: **31.03.2004**

(84) Designated Contracting States:
**AT BE BG CH CY CZ DE DK EE ES FI FR GB GR HU IE IT LI LU MC NL PL PT RO SE SI SK TR**
Designated Extension States:
**AL LT LV MK**

(71) Applicant: **Nederlandse Organisatie voor Toegepast-Natuurwetenschappelijk Onderzoek TNO**
**2628 VK Delft (NL)**

(72) Inventor: **Berkhoff, Arthur Perry**
**7006 EY Doetinchem (NL)**

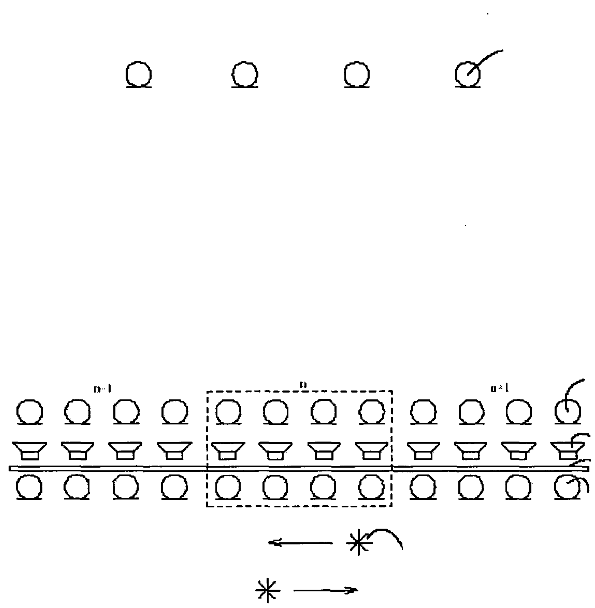
(74) Representative:
**Winckels, Johannes Hubertus F. et al**
**Vereenigde**
**Johan de Wittlaan 7**
**2517 JR Den Haag (NL)**

(54) **System for actively reducing sound**

(57) A system for actively reducing sound from a primary noise source, such as traffic noise, comprising:

a loudspeaker connector for connecting to at least one loudspeaker for generating anti-sound for reducing said noisy sound;
a microphone connector for connecting to at least a first microphone placed adjacent to said loudspeaker;
a control unit coupled to said first microphone connector, for providing an error signal, based on the output of said first microphone; and
a control unit for outputting a signal to said loudspeaker connector, for controlling said loudspeaker based on said error signal of said control unit. According to the invention, said control unit is provided with a processor adapted to provide said error signal as a simulated error signal of a virtual microphone placed in the far-field of the loudspeaker. In this way a better noise suppression in the far-field can be achieved.

Figure 1



EP 1 583 075 A1

## Description

**[0001]** The invention relates to a system for actively reducing sound from a primary noise source, such as traffic noise, comprising: a loudspeaker connector for connecting to at least one loudspeaker for generating anti-sound for reducing said noisy sound; a microphone connector for connecting to at least a first microphone placed adjacent to said loudspeaker; a control unit coupled to said first microphone connector, for providing an error signal, based on the output of said first microphone; and a control unit for outputting a signal to said loudspeaker connector, for controlling said loudspeaker based on said error signal of said control unit.

**[0002]** Recently, such systems have been considered as active noise barriers for reduction of traffic noise. The goal of the active noise barriers is to reduce the noise on positions distant from the noise barrier, such as on the facades of houses. Microphones at these positions can be used in order to create error signals to be minimized by the controller. A problem with this is the vulnerability of connections and microphones and a therefore expected short economic life. In another configuration, microphones were placed directly near the active noise barrier. This leads to obvious advantages due to the absence of sensors at inconvenient positions. This can lead to a possibly more robust system with respect to fluctuations of the wind. However, it has been found that the minimization of sound near the loudspeakers is not always appropriate for reducing the sound pressure at more distant locations.

**[0003]** The invention has as one of its goals to provide a system that at the one time offers a compact and robust way of implementing a noise barrier but where the hereabove described disadvantages are mitigated.

**[0004]** This is achieved by providing a system according to the features of claim 1.

**[0005]** Conveniently, it has been found that by deriving a simulated error signal of a virtual microphone placed in the far-field of the loudspeaker, the noise suppression is much more successful than conventional active noise barriers and offers reductions even up to 10 dB or more compared to a sound reduction of a conventional system with adjacently placed microphones.

**[0006]** Since the primary and secondary sources may have a very different nature they may lead to rather different requirements for a transfer function. One big difference, for example, is that the nearfield sensors are located rather far away from the primary sources while the nearfield sensors are, by definition, in the nearfield of the secondary sources. So, although a single transfer function Hyz may have acceptable performance in some systems, such as active panels, the performance for active noise barriers can be improved further. Therefore, in a further preferred embodiment, said simulated error signal of said virtual microphone is a result of a combination of a first transfer function expressing the far field character of the primary source near said first microphone and a second transfer function expressing the near field character of the secondary source near said first microphone.

**[0007]** The loudspeaker may be arranged to produce anti-sound in a direction away from said primary noise source. Further, said microphone is preferably placed adjacent in front of said loudspeaker.

**[0008]** In a still further preferred embodiment, a second microphone may be placed between said loudspeaker and said primary source. Said second microphone may be arranged to provide a second error signal that is used as a feed forward error signal in order to achieve an expected sound pressure level of the primary source near said loudspeaker.

**[0009]** Said at least one loudspeaker and said at least first microphone may be part of an array of loudspeakers and an array of microphones respectively, wherein loudspeakers and said microphones are placed at a distance less than 5 times an interspacing between two adjacent loudspeakers. Such an array may serve as an active noise barrier on the side of a road etc. Still more preferably, said at least said at least one loudspeaker and said at least one microphone are part of an array of loudspeakers and an array of microphones respectively, wherein said loudspeakers and said microphones are placed relative to each other in a range between 10% and 100% of an interspacing between two adjacent loudspeakers. It has been found that a position of said microphone at such a general "close" distance of the loudspeaker, is still able to derive a sufficient error signal in order to be able to calculate a far field sound pressure.

**[0010]** As a practical embodiment, said microphone may be formed integral with said loudspeaker in a panel to be placed on the side of a road. To this end, the invention is also related to a panel for a noise screen to be placed on the side of the road, for actively reducing noise from a primary source, comprising: a loudspeaker to be directed away from the primary source

a microphone attached to said panel and placed on a distance away from said panel; and a system according to any of the preceding aspects.

**[0011]** Furthermore, said panel may further comprise a second microphone placed opposite to said first microphone, viewed in a direction away from said panel. Furthermore, said loudspeaker is placed on top of said panel. In this way the influence of wind is more or less the same for the primary sources and the secondary sources. This can lead to a more robust system with respect to fluctuations of the wind.

**[0012]** In another aspect, the invention is related to a method as described in claims 12 and 13.

**[0013]** The invention will be further clarified with reference to the figures. In the figures:

Figure 1 shows a basic configuration according to an embodiment of the invention;

Figure 2, 3 and 4 show prior art system diagrams wherein noise control is actively used;
Figure 5 is a schematic system diagram of the system according to the invention;
Figure 6 shows a sectional diagram of Figure 5 explaining the invention in more detail;
Figure 7 shows a configuration of an active noise control system for reduction of noise at the position of the farfield sensors by using information derived from nearfield error sensors showing the region for identification related to the primary noise sources;
Figure 8 shows a comparison of a primary farfield signal with an estimate obtained from the nearfield sensor signals;
Figure 9 shows a comparison of a secondary transfer function to the farfield with an estimate obtained from the nearfield sensor signals;
Figure 10 shows a reduction of the broadband sound pressure for three independent broadband primary noise sources at z= -4 m; the error signal equals the nearheld pressure;
Figure 11 shows a reduction as in Figure 10; the error signal equals a measured farfield pressure;
Figure 12 shows a reduction as in Figure 10; the error signal equals a virtual farfield pressure.

**[0014]** In the description, like or corresponding elements in will be referenced with the same reference numerals.

**[0015]** A proposed configuration of the system according to the invention can be found in Figure 1. In this figure reference sensors 1 (microphones) are present near secondary sources 2 (loudspeakers), and nearfield error sensors 3 (microphones). In an initial calibration phase also farfield error sensors 4 (microphones) can be used. One section (n) of the active noise barrier 5 is considered. In order to assess the stability and performance of the active noise barrier 5, including the interactions between different subsystems(n-1, n+1), also the sensors and sources of adjacent sections should be taken into account.

**[0016]** One of the aspects of the system is that virtual error signals are derived from the nearfield error sensors. These virtual error signals should represent the degrees of freedom in the farfield of the angular sector of interest. In the configuration, we distinguish the following components: the primary disturbance *d*, reference signals *x*, actuator control signals *u*, nearfield error signals *e_y*, and farfield error signals *e_z* such that

$$\begin{bmatrix} x \\ e_y \\ e_z \end{bmatrix} = \begin{bmatrix} G_{dx} & G_{ux} \\ G_{dy} & G_{uy} \\ G_{dz} & G_{uz} \end{bmatrix} \begin{bmatrix} d \\ u \end{bmatrix} \qquad (1)$$

**[0017]** The action of the control operator W is described by

$$u = Wx \qquad (2)$$

**[0018]** Lower case variables denote vectorial quantities that are a function of time, whereas upper case variables denote matrices that operate on the time-dependent vectors. The assumption is that $G_{ux}$ is compensated in the controller, leading to a control structure based on Internal Model Control. Furthermore, aspects related to decentralization are not taken into account yet but will be discussed in the remainder.

**[0019]** For purposes of comparison, some existing systems will be described with reference to the Figure 2 -4. Figure 2 shows a system using a single transfer function $H_{yz}$. Such systems are used for actively controlling noise from panels. However, the performance for the present application is insufficient, as can be demonstrated in simulations. Figure 3 shows a system where the basic strategy is to minimize the nearfield pressure, using a feedforward controller, as in Figure 2. Figure 4 shows a system in which the nearfield pressure is directly fed back to the controller, where the basic strategy is also to minimize the nearfield pressure. The latter configuration has been tested by Japanese researchers. In the present application, reduction of the nearfield pressure is very different from minimizing the farfield pressure. The simulations described below will show that the configuration of Figure 5 performs substantially better than the configurations of Figure 2-4.

**[0020]** Turning to Figure 5, in a first approach for the reduction of farfield sound pressure the sound pressure is reduced near the noise barrier. This is the approach that is used in, for example, a Japanese active noise barrier as illustrated in Figure 4. A disadvantage of this system is that a reduction of the nearfield pressure not necessarily leads to a reduction of the farfield pressure. This holds for both feedback configurations and for feedforward configurations. According to the invention, an improved technique is based on the estimation of the farfield error signals *e_z* (which are assumed to be unavailable when the system is operational) from the nearfield error signals *e_y*.

**[0021]** In a first approach a transfer function $H_{yz}$ is designed which makes a real-time estimate of *e_z* from measured data $e\_y$. However, it was found that the maximum performance with such a transfer function is limited by the fact that we are dealing with two sets of sources, the primary sources and the secondary sources. These two sets of sources may have a very different nature and may lead to rather different requirements for the transfer function $H_{yz}$. One big difference, for example, is that the nearfield sensors are located rather far away from the primary sources while the nearfield sensors are, by definition, in the nearfield of the secondary sources. So, although a single transfer function $H_{yz}$ may have acceptable performance in some systems, such as active panels, the performance for active noise barriers can be improved further.

**[0022]** In Figure 5 and Figure 6 is illustrated, that, for further improving the noise control in the far field, a new strategy was devised in which the primary sources and the secondary sources were treated separately. This estimation proceeds as follows. The actual contributions on $e\_y$ and $e\_z$ consist of a part due to the primary signal and a part due to the secondary signal. As the transfer functions from the primary source and secondary source to both $e\_y$ and $e\_z$ are quite different, these contributions are taken into account separately. Therefore, we define $G_{dy}$ as the transfer function between primary source and nearfield error sensor, $G_{dz}$ as the transfer function between primary source and farfield error sensor, $G_{uy}$ as the transfer function between secondary source and nearfield error sensor, and $G_{uz}$ as the transfer function between secondary source and farfield error sensor. The farfield signal $e\_z$ can be expressed as

$$e\_z = d\_z + s\_z \qquad (3)$$

in which the primary signal $d\_z$ is given by

$$d\_z = G_{dz}\, d \qquad (4)$$

and the secondary signal s_z is given by

$$s\_z = G_{uz}\, u \qquad (5)$$

**[0023]** The estimate $d\hat{}z$ of the signal $d\_z$ is derived from an estimate $d\hat{}y$ of the primary nearfield error signals as follows

$$d\hat{}z = H_{yz}\{d\}\, d\hat{}y \qquad (6)$$

where $H_{yz}\{d\}$ is obtained from system identification. The estimate $d\hat{}y$ is obtained by subtracting the secondary signal from the measured signal $e\_y$:

$$d\hat{}y = e\_y - s\hat{}y \qquad (7)$$

where $s\hat{}y$ is an estimate of the secondary signal on the nearfield error sensors, which is given by

$$s\hat{}y = G\hat{}_{uy}\, u \qquad (8)$$

**[0024]** An estimate of the contribution of the secondary sources on the farfield sensors is given by

$$s\hat{}z = H_{yz}\{s\}\, s\hat{}y \qquad (9)$$

where $H_{yz}\{s\}$ is also obtained from system identification. Finally, the estimate $e\hat{}z$ of the farfield sensor can be obtained from the summation of the estimates of the primary sources and secondary sources:

$$e\hat{}z = d\_z + s\_z \qquad (10)$$

**[0025]** The estimate e^z can be written as

$$\hat{e}_z = H_{yz}\{d\}\,\hat{d}_y + H_{yz}\{s\}\,\hat{s}_y$$
$$= H_{yz}\{d\}\,e\_y + (H_{yz}\{s\} - H_{yz}\{d\})\hat{G}_{uy}\,u \qquad (11)$$

**[0026]** A block diagram of the above procedure is given in Figure 6. The nearfield error signal is given by

$$e\_y = d\_y + s\_y \qquad (12)$$

**[0027]** Therefore, using

$$d\_y = G_{dy}\,d \qquad (13)$$

and

$$s\_y = G_{uy}\,u \qquad (14)$$

the estimate e^z can be written as

$$e\hat{\ }z = H_{yz}\{d\}\,(G_{dy}d + G_{uy}\,u) + (H_{yz}\{s\} - H_{yz}\{d\})G\hat{\ }_{uy}\,u \qquad (15)$$

**[0028]** In case of perfect modeling we have $G_{uy}$ = G^uy. Then the above expression reduces to

$$e\hat{\ }z = H_{yz}\{d\}\,G_{dy}\,d + H_{yz}\{s\}\,G_{uy}\,u \qquad (16)$$

**[0029]** If the secondary sources are close to the diffraction sources and if they radiate in a similar way then we have $H_{yz}\{d\} = H_{yz}\{s\} = H_{yz}$, leading to

$$e\hat{\ }z = H_{yz}(G_{dy}\,\{d\} + G_{uy}\{u\}) = H_{yz}\,e\_y \qquad (17)$$

**[0030]** Stability robustness and performance robustness are governed by the sensitivity of the control system to changes in the secondary path. The propagation distances involved in the secondary path are considerably reduced if nearfield sensors are used instead of the farfield sensors. Therefore, it is expected that the reduced change of the propagation path will have a positive effect on the stability robustness of the system. Furthermore, if the primary sources are effectively diffraction sources on top of a noise barrier and if the secondary sources are close to these diffraction sources then we obtain a system in which a large part of the possibly varying transmission path is common to both the primary signals and the secondary signals. This results in an expected improved performance robustness with respect to changes in wind, temperature, etc.

**[0031]** $E_q$. (17) suggests that minimizing *e_y* would be sufficient to reduce *e^z.* This only holds if the primary sources exactly coincide with the secondary sources. In practical situations it is very difficult to exactly realize the condition $H_{yz}\{d\} = H_{yz}\{s\} = H_{yz}$. Differences are caused by, for example, the distributed nature of the primary (diffraction) sources while the secondary sources are concentrated sources. These different behaviors are especially seen on the nearfield sensors. Also the exact position of the primary sources may be difficult to determine. Therefore, for a practical implementation a more general approach is required, such as based on the scheme of Figure 6 where the estimation of the primary path and the estimation of the secondary path are treated separately. For the estimation of the primary transfer functions we define a region with identification sources as shown in Figure 7.

**[0032]** Furthermore, only those (and precisely those) nearfield pressure distributions should be controlled that contribute to the farfield, as governed by the transfer functions $H_{yz}$. If *e_y* is minimized directly then usually only sound pressure reductions in the nearfield are obtained. An additional reduction of the order of $H_{yz}$ can lead to systems that are potentially more robust, have less spillover and require less actuator effort.

**[0033]** In order to be able to compute the virtual error signals an initial calibration phase is assumed in which sources are placed at random positions in the region where the primary noise sources are to be expected. The estimate of the primary signal on one of the farfield sensors is shown in Figure 8. The estimate of one of the secondary source transfer functions to the farfield as estimated from the secondary transfer functions to the nearfield sensors is shown in Figure 9. In an example 50 calibration sources have been used, which are indicated as green asterixes in Figure 12. Three independent primary sources are used, also shown in Figure 12. The identification procedure is based on the solution of the equations resulting from an assumed multi-input, multi-output Finite Impulse Response model leading to a block-Toeplitz structure. In order to improve the stability of the solution, the mean-square value of the coefficients of the FIR-model are weighted with a normalized coefficient weighting factor $\beta = 10^{-2}$ as compared to the mean-square prediction error of the farfield pressure. This value of $\beta$ was obtained by reducing the value of $\beta$ until the prediction error for primary signal validation data did not reduce anymore. The latter prediction error was obtained with validation data from an independent set of primary sources at random positions in the same region as for the solution of the system of equations.

**[0034]** Although the transfer functions $H_{yz}\{d\}$ and $H_{yz}\{s\}$ relate the signals from identical nearfield sensors and identical farfield sensors, they should be obtained from separate calibration procedures, as mentioned previously. The necessity of this is demonstrated in Table 1, which gives the estimation results on farfield sensor for different combinations of calibration sources and the two transfer functions $H_{yz}\{d\}$ and $H_{yz}\{s\}$. The configuration is identical to that in Fig. 12. It can be seen that the estimator $H_{yz}\{d\}$ only gives accurate results if the calibration sources are placed in the same region as where the primary sources are to be expected. The same holds for $H_{yz}\{s\}$, which is only accurate if the calibration sources are taken to be the secondary sources.

[Table 1]

| calibration sources | $H_{yz}\{d\}$ estimation error[dB] | $H_{yz}\{s\}$ estimation error[dB] |
|---|---|---|
| sources in primary source region | -14.6 | 10.5 |
| secondary sources | -2.6 | -36.3 |

**[0035]** The example of Figure 10 is an active noise barrier in which the error signal equals the pressure as measured on microphones near the secondary sources. In the present case this distance, i.e. the difference between the z-coordinate of the 5 secondary sources and z-coordinate of the 5 error sensors is 0.5 m. In addition, 5 reference sensors are used, which are positioned, as seen from the secondary sources, 1 m towards the primary sources. The primary noise sources are three independent broadband noise sources which are positioned at a distance of 4.5 m from the secondary sources. The position of the passive noise barrier could be at z=0 but for reasons of simplicity there is no such noise barrier. The sound pressure in the farfield is evaluated with 5 microphones at z = 20 m. The distance between these microphones is 1.5 m. The sampling frequency is f_s = 1 kHz, the number of controller coefficients equals 128, the distance between the secondary sources is 0.3 m, as are the distances between the reference sensors and the error sensors. All transducers and sources are at a height y = .2 m. Figure 10 shows the configuration and the resulting sound pressure in the x,z-plane. It can be seen that sound pressure reductions are mainly obtained near the error sensors. The reduction at the error sensors is 24.2 dB but the reduction in the farfield is much less, being 1.9 dB.

**[0036]** If the error sensors are moved to the farfield, viz. to the positions of the evaluation microphones, then the reductions in the farfield become much higher. The average reduction of the error signal and consequently the reduction at the evaluation microphones now becomes 11.3 dB. The results are shown in Figure 11.

**[0037]** Figure 12 shows the results in case the error signal equals virtual farfield signals, using the same microphone positions as in Figure 10. The virtual sensor signals are obtained by processing the nearfield error signals *e_y* with a fixed operator which is determined in a calibration phase. The sources as used in this calibration phase are positioned in the region where the primary noise sources are to be expected (see Figure 12). It can be seen that the resulting farfield sound pressure reductions are slightly less than in Figure 11, being 9.0 dB. However, the farfield sound pressure reductions are considerably higher than obtained with the nearfield error sensors as used in Figure 10. The reduction of the error signals is approximately equal in case of minimizing the virtual error signals, being 11.6 dB, and in case of minimizing the true farfield error signals (11.3 dB). The numerical results for the three cases are collected in Table 2.

[Table 2]

| Error signal | error signal av. reduction [dB] | farfield pressure av. reduction [dB] |
|---|---|---|
| Figure 10 | 24.2 | 1.9 |
| Figure 11 | 11.3 | 11.3 |
| Figure 12 | 11.6 | 9.0 |

Robustness of the controllers was evaluated for the following cases: for a different order of the controller, for a new set of primary sources in the same region as for the nominal case, for a change of the primary spectrum as compared to the nominal case, and for a moving source. It was found that a reduction of the number of controller coefficients leads to less reduction for the nominal case but also to less performance degradation for changing primary source distributions. Similar trends were found for the case of 32 controller coefficients. Hence, as compared to the controllers with 128 coefficients, the performance of the systems with 32 and 64 coefficients is less dependent on the actual primary source distribution and is therefore more stable and predictable. The influence of a simultaneous change of the positions of the primary sources and a changing primary spectrum was investigated using a modified primary spectrum that was obtained by applying a fourth-order bandpass filter with Butterworth characteristic with lower and upper cutoff frequencies of 0.15 $f_s$ and 0.35 $f_s$, respectively, to the primary source signals. With the modified primary source positions and spectrum, a slight improvement was found of the performance on the farfield evaluation microphones from 10.6 dB to 11.4 dB. Furthermore, the performance for a moving source was investigated. The radiation characteristics of the moving sources were simulated by taking into account the doppler shift as well as the changing radiation characteristics depending on the direction of the movement. Also, simulations were performed in order to study the performance variations for a change in the height of the primary sources. The invention is not limited to the hereabove described embodiments but may comprise variations and modifications thereto while falling under the scope of the annexed claims.

## Claims

**1.** A system for actively reducing sound from a primary noise source, such as traffic noise, comprising:

- a loudspeaker connector for connecting to at least one loudspeaker for generating anti-sound for reducing said noisy sound;
- a microphone connector for connecting to at least a first microphone placed adjacent to said loudspeaker;
- a control unit coupled to said first microphone connector, for providing an error signal, based on the output of said first microphone; and
- a control unit for outputting a signal to said loudspeaker connector, for controlling said loudspeaker based on said error signal of said control unit; **characterized in that** said control unit is provided with a processor adapted to provide said error signal as a simulated error signal of a virtual microphone placed in the far-field of the loudspeaker.

**2.** A system according to claim 1, wherein said simulated error signal of said virtual microphone is a result of a combination of a first transfer function expressing the far field character of the primary source near said first microphone and a second transfer function expressing the near field character of the secondary source near said first microphone.

**3.** A system according to claim 1, wherein said loudspeaker is arranged to produce anti-sound in a direction away from said primary noise source.

**4.** A system according to claim 1, wherein said microphone is placed adjacent in front of said loudspeaker.

**5.** A system according to any of the preceding claims, wherein a second microphone is placed between said loudspeaker and said primary source.

**6.** A system according to claim 5, wherein said second microphone is arranged to provide a reference signal that is used as a feed forward signal in order to achieve an expected sound pressure level of the primary source near first microphone.

**7.** A system according to claim 1, wherein said at least one loudspeaker and said at least first microphone are part of an array of loudspeakers and an array of microphones respectively, wherein loudspeakers and said microphones are placed at a distance less than 5 times an interspacing between two adjacent loudspeakers.

**8.** A system according to claim 1, wherein said at least one loudspeaker and said at least one microphone are part of an array of loudspeakers and an array of microphones respectively, wherein said loudspeakers and said microphones are placed relative to each other in a range between 10% and 100% of an interspacing between two adjacent loudspeakers.

9. A system according to claim 1, wherein said microphone is formed integral with said loudspeaker in a panel to be placed on the side of a road.

10. A panel for a noise screen to be placed on the side of the road, for actively reducing noise from a primary source, comprising:

- a loudspeaker to be directed away from the primary source
- a microphone attached to said panel and placed on a distance away from said panel; and
- a system according to any of the claims 1-10.

11. A panel according to claim 10 further comprising a second microphone placed opposite to said first microphone, viewed in a direction away from said panel.

12. A panel according to claims 10-11 wherein said loudspeaker is placed on top of said panel.

13. A method of calibrating a system according to any of the claims 1-9 comprising:

- placing a panel according to claim 10 near the side of the road;
- placing a far field microphone at a distance away from said panel and said road;
- measuring an error signal derived from a first microphone placed adjacent to said loudspeaker;
- measuring a reference signal derived from the far field microphone;
- calibrating said system by a comparison between said error signal and said reference signal.

14. A method for actively reducing sound from a primary noise source, such as traffic noise, comprising:

- deriving an error signal, based on the output of a first microphone placed adjacent to a loudspeaker; and
- outputting a driving signal to said loudspeaker based on said error signal for generating anti-sound for reducing said noisy sound;
- wherein said error signal is a simulated error signal of a virtual microphone placed in the far-field of the loudspeaker.

15. A method according to claim 14, wherein said simulated error signal of said virtual microphone is obtained by combining a first transfer function expressing the far field character of the primary source near said first microphone and a second transfer function expressing the near field character of the secondary source near said first microphone.

Figure 1

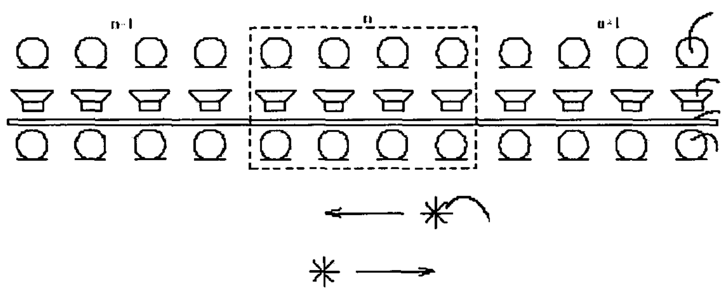


Figure 2

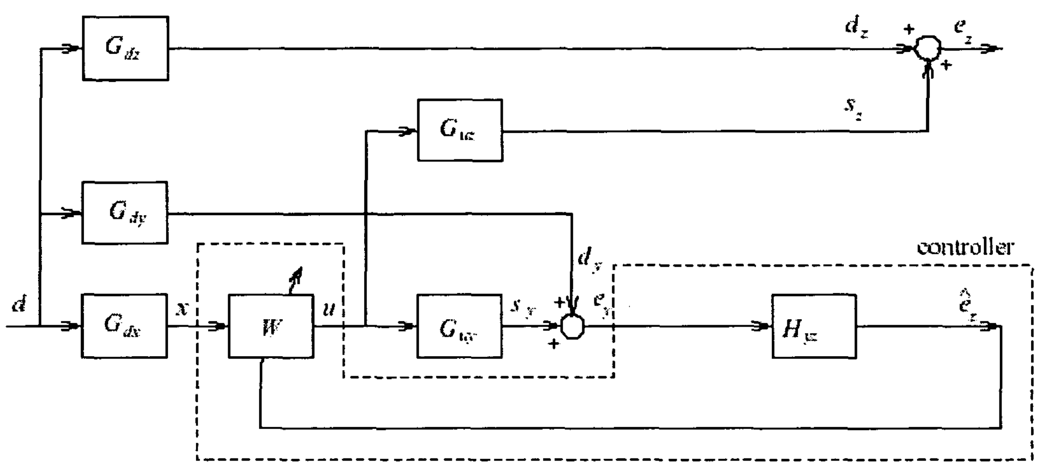



Figure 3

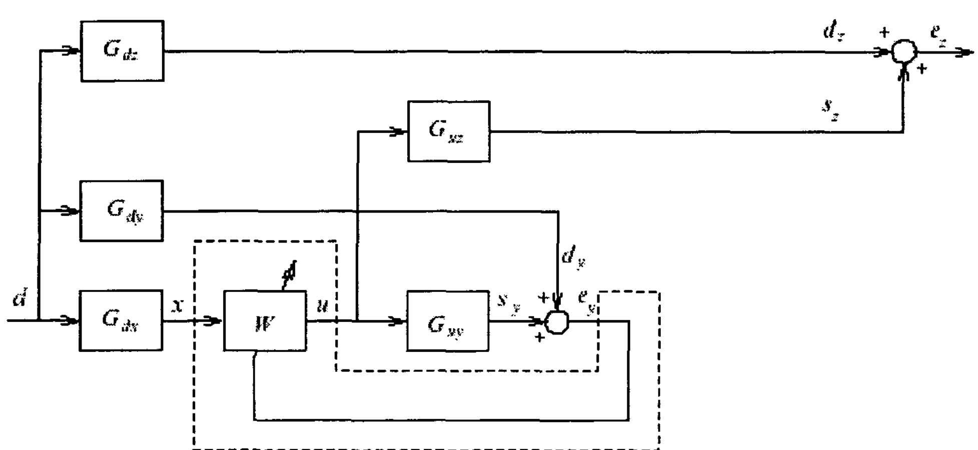


Figure 4

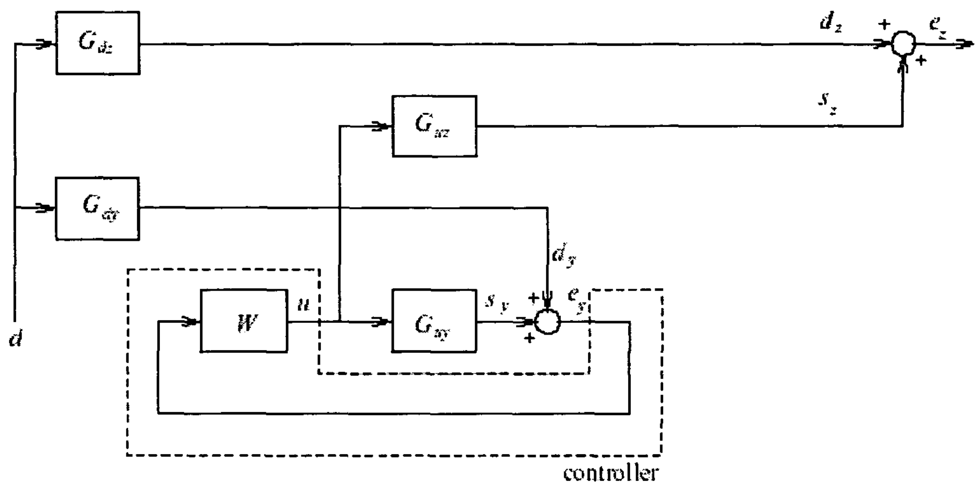



Figure 5

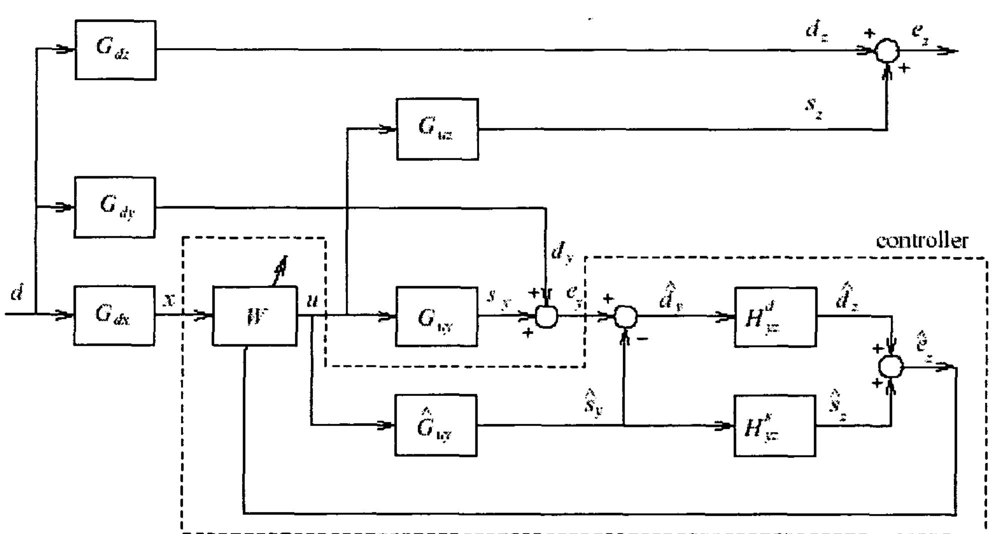


Figure 6

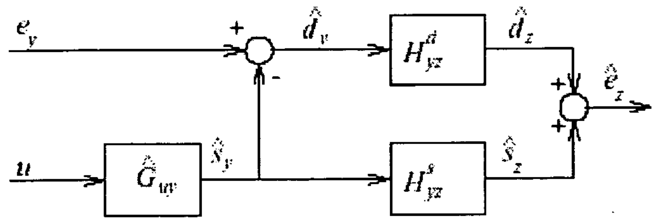



Figure 7

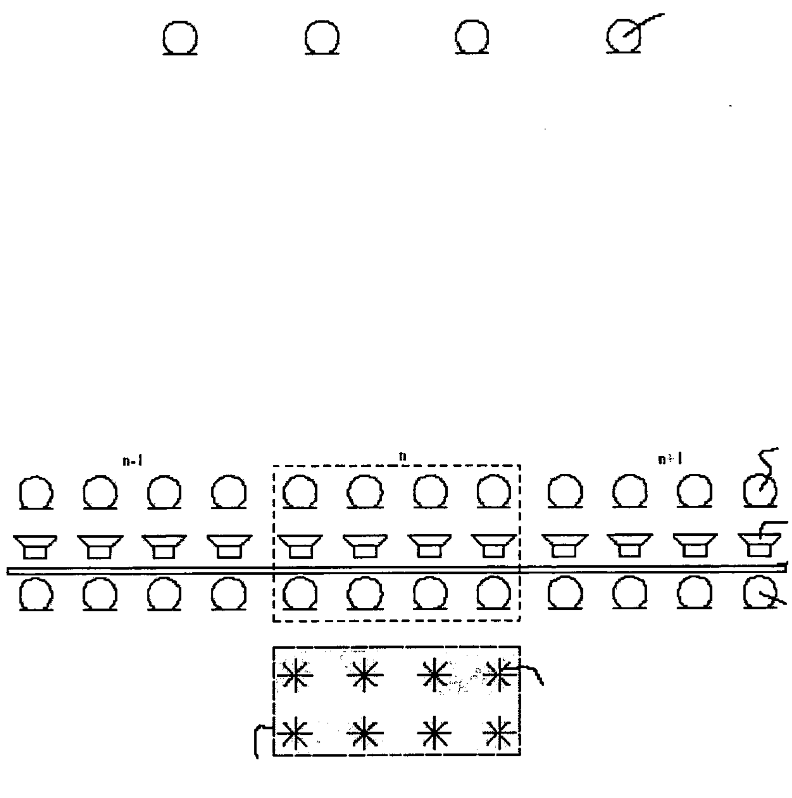


Figure 8

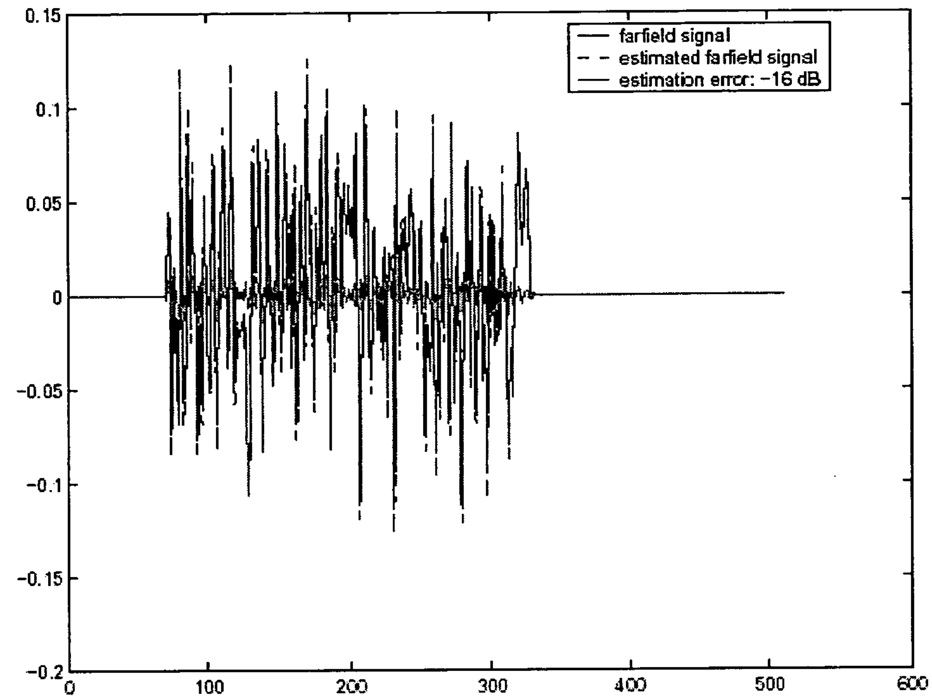



Figure 9

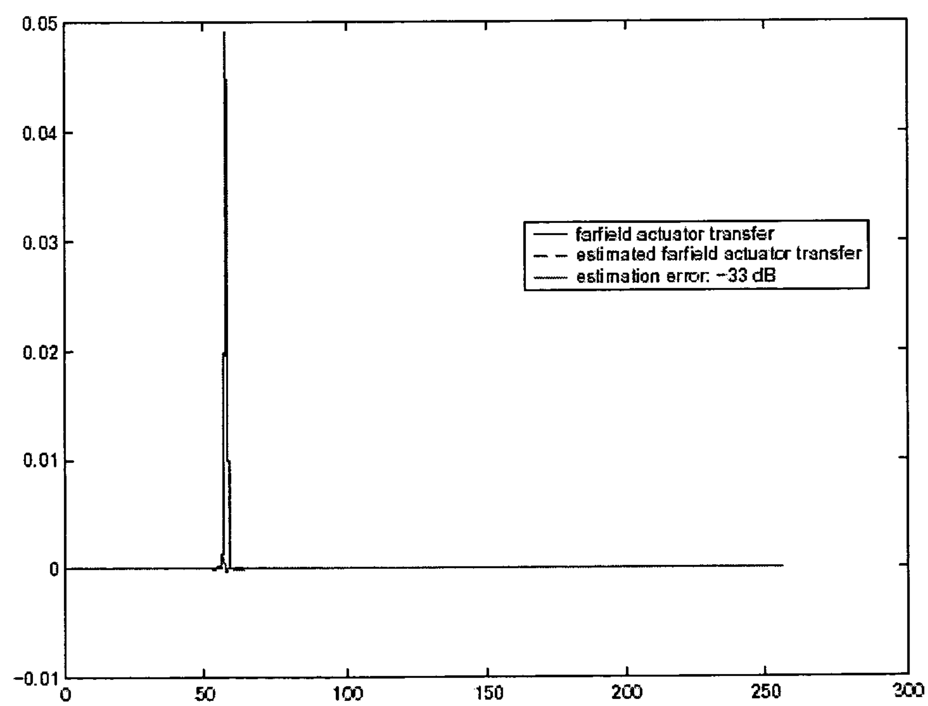


Figure 10

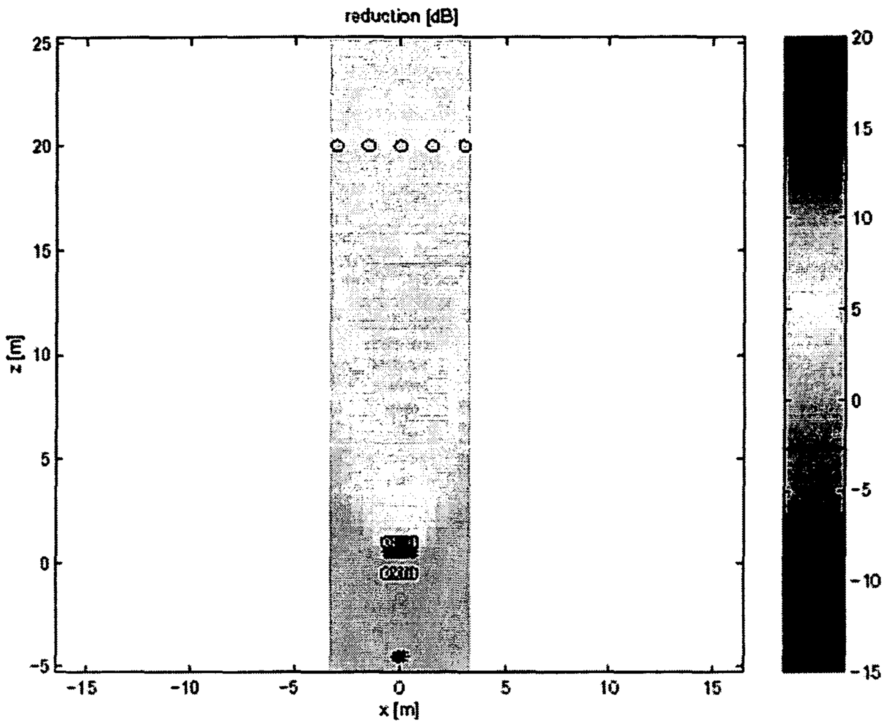



Figure 11

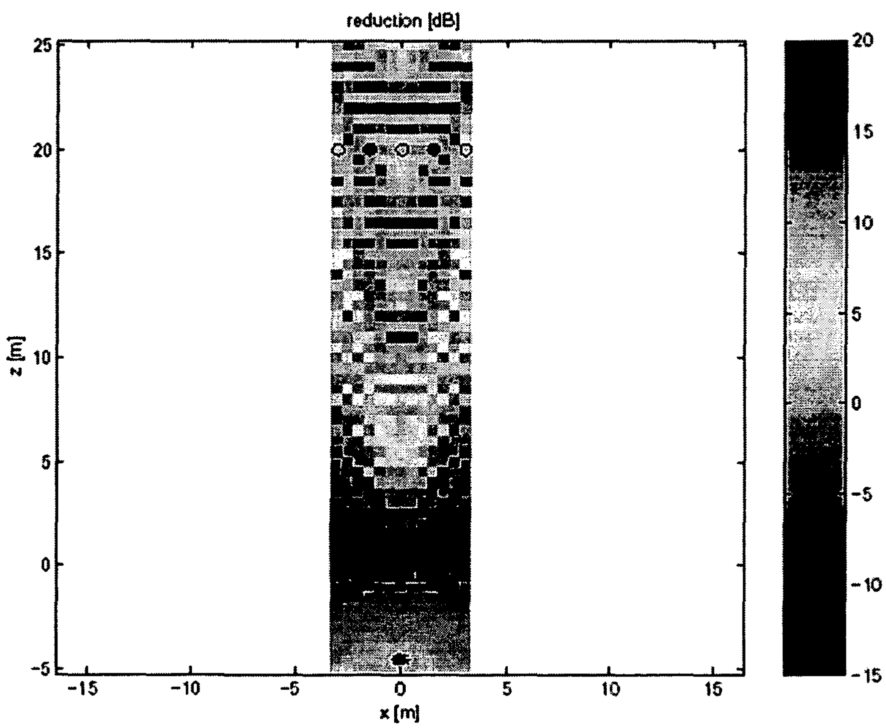
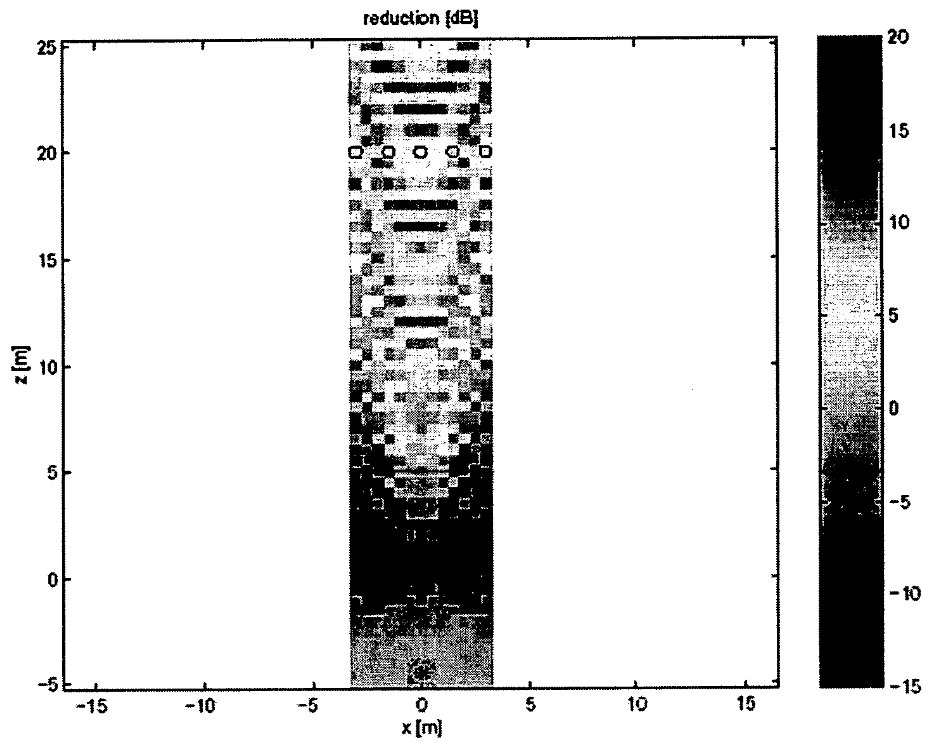


Figure 12


reduction [dB]
z [m]
x [m]
25
20
15
10
5
0
-5
-15
-10
-5
0
5
10
15
20
15
10
5
0
-5
-10
-15

European Patent Office

# EUROPEAN SEARCH REPORT

Application Number
EP 04 07 6031

## DOCUMENTS CONSIDERED TO BE RELEVANT

| Category | Citation of document with indication, where appropriate, of relevant passages | Relevant to claim | CLASSIFICATION OF THE APPLICATION (Int.Cl.7) |
|---|---|---|---|
| X<br><br><br><br><br><br><br><br><br>Y | GARCIA-BONITO J ET AL: "GENERATION OF ZONES OF QUIET USING A VIRTUAL MICROPHONE ARRANGEMENT"<br>JOURNAL OF THE ACOUSTICAL SOCIETY OF AMERICA, AMERICAN INSTITUTE OF PHYSICS. NEW YORK, US,<br>vol. 101, no. 6, 1 June 1997 (1997-06-01), pages 3498-3516, XP000696885<br>ISSN: 0001-4966<br>* the whole document * | 1-4, 13-15<br><br><br><br><br><br><br><br><br>5,6,9-12 | G10K11/178 |
| X | US 5 519 637 A (MATHUR GOPAL)<br>21 May 1996 (1996-05-21)<br>* abstract *<br>* column 2, line 66 - column 9, line 43 *<br>* figures 1-11 * | 1-4, 13-15 | |
| Y | EP 0 759 606 A (DIGISONIX INC)<br>26 February 1997 (1997-02-26)<br>* column 2, lines 33-43 *<br>* column 5, line 19 - column 7, line 3 *<br>* figures 1-21 * | 5,6 | |
| Y | US 2003/002687 A1 (KIESLINGER MICHAEL ET AL) 2 January 2003 (2003-01-02)<br>* page 3, paragraph 53-56 *<br>* figures 1,4 * | 9-12 | TECHNICAL FIELDS SEARCHED (Int.Cl.7)<br>G10K |
| A | US 5 315 661 A (EATWELL GRAHAM ET AL)<br>24 May 1994 (1994-05-24)<br>* the whole document * | 1-15 | |
| A | US 5 917 919 A (ROSENTHAL FELIX)<br>29 June 1999 (1999-06-29)<br>* column 5, line 29 - column 13, line 53 *<br>* figures 1-5 * | 1-15 | |

The present search report has been drawn up for all claims

| Place of search | Date of completion of the search | Examiner |
|---|---|---|
| The Hague | 17 September 2004 | Modesto, C |

CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
Y : particularly relevant if combined with another document of the same category
A : technological background
O : non-written disclosure
P : intermediate document

T : theory or principle underlying the invention
E : earlier patent document, but published on, or after the filing date
D : document cited in the application
L : document cited for other reasons

& : member of the same patent family, corresponding document

1

EPO FORM 1503 03.82 (P04C01)

## ANNEX TO THE EUROPEAN SEARCH REPORT ON EUROPEAN PATENT APPLICATION NO. EP 04 07 6031

This annex lists the patent family members relating to the patent documents cited in the above-mentioned European search report.
The members are as contained in the European Patent Office EDP file on
The European Patent Office is in no way liable for these particulars which are merely given for the purpose of information.

17-09-2004

| Patent document cited in search report | | Publication date | Patent family member(s) | | | Publication date |
|---|---|---|---|---|---|---|
| US 5519637 | A | 21-05-1996 | NONE | | | |
| EP 0759606 | A | 26-02-1997 | US | 5715320 | A | 03-02-1998 |
| | | | CA | 2183669 | A1 | 22-02-1997 |
| | | | DE | 69626902 | D1 | 30-04-2003 |
| | | | DE | 69626902 | T2 | 21-08-2003 |
| | | | EP | 0759606 | A2 | 26-02-1997 |
| US 2003002687 | A1 | 02-01-2003 | AU | 770088 | B2 | 12-02-2004 |
| | | | AU | 5690100 | A | 30-05-2001 |
| | | | BR | 0015585 | A | 09-07-2002 |
| | | | CA | 2388179 | A1 | 25-05-2001 |
| | | | CN | 1390346 | T | 08-01-2003 |
| | | | DE | 10085355 | T0 | 15-05-2003 |
| | | | EP | 1230637 | A1 | 14-08-2002 |
| | | | WO | 0137256 | A1 | 25-05-2001 |
| | | | GB | 2370940 | A ,B | 10-07-2002 |
| | | | JP | 2003514265 | T | 15-04-2003 |
| US 5315661 | A | 24-05-1994 | AT | 194040 | T | 15-07-2000 |
| | | | CA | 2142013 | A1 | 03-03-1994 |
| | | | DE | 69231190 | D1 | 27-07-2000 |
| | | | DE | 69231190 | T2 | 22-03-2001 |
| | | | EP | 0657058 | A1 | 14-06-1995 |
| | | | JP | 8500193 | T | 09-01-1996 |
| | | | WO | 9405005 | A1 | 03-03-1994 |
| US 5917919 | A | 29-06-1999 | NONE | | | |



For more details about this annex : see Official Journal of the European Patent Office, No. 12/82