



(51) International Patent Classification:

G01R 33/28 (2006.01) H04B 1/12 (2006.01)
G10L 21/02 (2006.01) H04B 15/00 (2006.01)

(21) International Application Number:

PCT/GB2009/002634

(22) International Filing Date:

6 November 2009 (06.11.2009)

(25) Filing Language:

English

(26) Publication Language:

English

(30) Priority Data:

0820653.4 11 November 2008 (11.11.2008) GB

(71) Applicant (for all designated States except US): **ISIS INNOVATION LIMITED**; Ewert House, Ewert Place, Summertown, Oxford OX2 7SG (GB).

(72) Inventors; and

(75) Inventors/Applicants (for US only): **KOCHANSKI, Gregory, Peter** [US/GB]; Oxford University Phonetics Laboratory, 41 Wellington Square, Oxford OX1 2JF (GB). **DOBSON, Dermot, Brian** [GB/GB]; 93 Old Road, Headington, Oxford OX3 8SX (GB).

(74) Agents: **MERRYWEATHER, Colin, Henry** et al.; J. A. Kemp & Co., 14 South Square, Gray's Inn, London WC1R 5JJ (GB).

(81) Designated States (unless otherwise indicated, for every kind of national protection available): AE, AG, AL, AM, AO, AT, AU, AZ, BA, BB, BG, BH, BR, BW, BY, BZ, CA, CH, CL, CN, CO, CR, CU, CZ, DE, DK, DM, DO, DZ, EC, EE, EG, ES, FI, GB, GD, GE, GH, GM, GT, HN, HR, HU, ID, IL, IN, IS, JP, KE, KG, KM, KN, KP, KR, KZ, LA, LC, LK, LR, LS, LT, LU, LY, MA, MD, ME, MG, MK, MN, MW, MX, MY, MZ, NA, NG, NI, NO, NZ, OM, PE, PG, PH, PL, PT, RO, RS, RU, SC, SD, SE, SG, SK, SL, SM, ST, SV, SY, TJ, TM, TN, TR, TT, TZ, UA, UG, US, UZ, VC, VN, ZA, ZM, ZW.

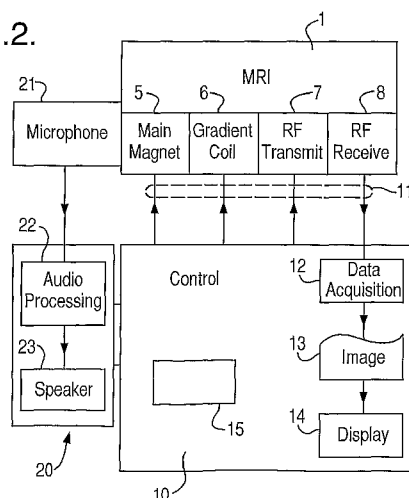
(84) Designated States (unless otherwise indicated, for every kind of regional protection available): ARIPO (BW, GH, GM, KE, LS, MW, MZ, NA, SD, SL, SZ, TZ, UG, ZM, ZW), Eurasian (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European (AT, BE, BG, CH, CY, CZ, DE, DK, EE, ES, FI, FR, GB, GR, HR, HU, IE, IS, IT, LT, LU, LV, MC, MK, MT, NL, NO, PL, PT, RO, SE, SI, SK, SM, TR), OAPI (BF, BJ, CF, CG, CI, CM, GA, GN, GQ, GW, ML, MR, NE, SN, TD, TG).

Published:

- with international search report (Art. 21(3))
- before the expiration of the time limit for amending the claims and to be republished in the event of receipt of amendments (Rule 48.2(h))

(54) Title: ACOUSTIC NOISE PREDICTION AND SUBTRACTION FROM INTERCOM AUDIO SIGNAL DURING MAGNETIC RESONANCE IMAGING

Fig.2.



(57) Abstract: In an intercom system for transmitting sound from a patient microphone disposed inside the MRI apparatus to an operator loudspeaker disposed outside the MRI apparatus, the acoustic noise generated by the MRI apparatus is predicted from gradient monitor signals representative of the drive signals applied to the gradient coils of the MRI apparatus, and is subtracted from the transmitted audio signal. The drive signals applied to the gradient coils are good noise predictors, allowing effective noise reduction. Also, the gain of the intercom system is adjusted to be higher when it is determined that the MRI apparatus is in an operational state of generating an MRI image using the gradient coils of the MRI apparatus.

ACOUSTIC NOISE PREDICTION AND SUBTRACTION FROM INTERCOM AUDIO SIGNAL DURING MAGNETIC RESONANCE IMAGING

The present invention relates to MRI (Magnetic Resonance Imaging) and specifically to dealing with the acoustic noise generated by an MRI apparatus during operation in an intercom system that transmits audio signals between a patient being imaged and an operator.

MRI apparatuses generate large amounts of acoustic noise during operation, to the extent that patients being imaged need to wear ear-protectors and/or ear-plugs.

Often an MRI apparatus will have an intercom system allowing communication between the patient and the operator. The acoustic noise generated by the MRI apparatus hinders the use of the intercom system. The acoustic noise is picked up by a microphone arranged in the MRI apparatus and will mask the patient's voice within the audio signal transmitted to the operator, in practice making it nearly impossible to hear anything the patient says when the MRI apparatus is operating to derive an MRI image.

Such difficulties in communicating with the patient, as well as hindering the practicalities of image acquisition, can have a negative psychological impact on the patient. In practice the overall experience of MRI imaging is psychologically difficult for a substantial number of patients who are uncomfortable in the narrow enclosed tube, this being exacerbated by the realisation that communication with the radiologist is difficult. This is a significant problem in practice with some patients, perhaps of the order of 5%, being unable to tolerate the MRI imaging at all.

Some methods and systems which implement reduction of acoustic noise generated by a MRI apparatus in an audio signal transmitted by an intercom system are known.

Several documents, namely US-4,696,030; NessAiver et al. "Recording high-quality speech during tagged cine-MRI studies using a fiber-optic microphone", J. Magnetic Resonance Imaging 2006, 23(5) 783, doi:10.1002/jmri.20463; and Cusak et al. "Automated Post-Hoc Noise Cancellation Tool for Audio Recordings Acquired in an MRI scanner", doi:10.1002/hbm.20085, Human Brain Mapping (2005) 24: 299-3, disclose techniques in which the acoustic noise of an MRI apparatus is recorded when no speech signal is present and is then subsequently subtracted from an audio signal received from a microphone when a patient might be speaking. Such systems rely on the perfect reproducibility of the acoustic noise which is a difficult to implement in practise with sufficient precision to provide effective noise reduction.

Another possibility in principle might be to provide an additional microphone and to predict the acoustic noise from the signal output thereby for subtraction from the audio signal transmitted by an intercom system from a main patient microphone to an operator loudspeaker, for example applying a noise-cancelling microphone of the type disclosed in US-6,963,649. Such prediction may be achieved by convolution with a convolution kernel. However, the degree of noise reduction available would be modest and there are practical problems. The achievable noise reduction is intrinsically limited by the separation of the microphones, which is in turn limited by the distance of the pair from the patient.

There are two competing constraints. The first constraint is that it is desirable for the microphones to be close together as possible so as to both receive sound from the same directions. Otherwise, sound waves propagating in different directions will experience a different time delay between the microphones, and as a result of the two microphones having different pick-up patterns, sound waves will be picked up by the two microphones at different amplitudes, depending on the direction of propagation, which is unknown, with the effect that the predicted noise can not perfectly match the actual noise. The second constraint is that the patient's speech be much louder in one microphone than the other. Absent this constraint, the same signal processing that reduces noise will also cancel out the patient's speech. Essentially, the system distinguishes between desirable speech and noise in that the speech is asymmetrical between the pair of microphones and the noise is symmetrical. It is not possible to satisfy both constraints at once, and this physical limitation limits noise reduction performance.

Also, in practice the microphones need to be very close to the patient's mouth, say within 5 cm, making the microphone assembly inconvenient to position and making cleanliness more problematic.

According to a first aspect of the present invention, there is provided a method of reducing acoustic noise generated by a MRI apparatus in an audio signal transmitted by an intercom system from a patient microphone disposed inside the MRI apparatus to an operator loudspeaker disposed outside the MRI apparatus, the method comprising:

- deriving gradient monitor signals representative of the drive signals applied to the gradient coils of the MRI apparatus;

- predicting the acoustic noise generated by the MRI apparatus from prediction signals including at least the gradient monitor signals; and

- subtracting the predicted acoustic noise from the audio signal transmitted by the intercom system.

Further according to the first aspect of the present invention, there is provided an intercom system in which a similar method is implemented.

Thus, reduction of acoustic noise generated by a MRI apparatus in an audio signal transmitted by an intercom system is achieved by using the drive signals applied to the gradient coils of the MRI apparatus as a predictor of the actual noise. It has been appreciated that this technique allows one to achieve effective noise reduction because those drive signals are very effective predictors. This is because the acoustic noise is generated predominantly by the physical processes resulting from current flow in the gradient coils. As the gradient coils are immersed in the field of the powerful main magnet, any current flow in the gradient coils yields large forces on the coils resulting in vibration of the structure of the MRI apparatus generating the main component of acoustic noise. As a result of this causation of the acoustic noise, the drive signals applied to the

gradient coils are highly effective predictors, allowing effective noise reduction.

In particular, as compared to the proposal discussed above of deriving a noise prediction using an additional microphone, the first aspect of the present invention does not suffer from the problems discussed above with the effect that it is possible to achieve more effective noise reduction.

The noise reduction achieved by the first aspect of the present invention improves the quality of communication between the patient and the operator and thereby also reduces the psychological impact of the MRI imaging on the patient.

According to a second aspect of the present invention, there is provided a method of automatically controlling the gain of an intercom system of an MRI apparatus transmitting an audio signal from a patient microphone disposed inside the MRI apparatus to an operator loudspeaker disposed outside the MRI apparatus, the method comprising:

determining whether or not the MRI apparatus is in an operational state of generating an MRI image using the gradient coils of the MRI apparatus; and

controlling the gain of the intercom system to apply a lower gain when it is determined that the MRI apparatus is in said operational state than when it is determined that the MRI apparatus is not in said operational state.

Further according to the second aspect of the present invention, there is provided an intercom system in which a similar method is implemented.

This second aspect of the invention therefore provides automatic control of the gain of the intercom system in response to whether the MRI apparatus is in an operational state of generating an MRI image using the gradient coils of the MRI apparatus, this being of course the operational state in which loud acoustic noise is generated. In particular the gain is reduced when acoustic noise is generated and increased when acoustic noise is not generated. The reason for this is that any acoustic noise on the transmitted audio signal while the MRI apparatus is operating can cause the operator to reduce the gain manually to a level at which the noise is not unpleasant or annoying. In the absence of the present invention, when operation of the MRI apparatus subsequently ceases this can have the effect that the low gain lowers the audibility of the audio signal when the patient speaks, unless and until the operator increases the gain. In practice this can interrupt conversation between the patient and the operator, often requiring the operator to manually re-adjust the gain.

However, the second aspect of the present invention reduces this problem by automatically having a higher gain when the MRI apparatus is not operating. Thus the patient is more audible in this state without the need for the operator to manually increase the gain.

To allow better understanding, an embodiment of the present invention will now be described by way of non-limitative example with reference to the accompanying drawings, in which:

-4-

Fig. 1 is a schematic cross-sectional view of an MRI apparatus;

Fig. 2 is a diagram of the circuitry of the MRI apparatus;

Fig. 3 is a circuit diagram of an intercom system;

Fig. 4 is a diagram of a pick-up circuit for gradient monitor signals;

Fig. 5 is a block diagram of the audio processing block of the intercom system; and

Fig. 6 is a diagram of an RF pick-up circuit.

An MRI apparatus 1 for imaging a patient 2 is shown in Fig. 1 and has a conventional construction.

The MRI apparatus 1 has a table 3 which carries the patient 2 into a tunnel 4 inside the bore of a main magnet 5. In operation the main magnet 5 generates a high magnetic field inside the tunnel 4. The main magnet 5 may be a permanent magnet, a resistive magnet, or a superconducting magnet in which case the MRI apparatus 1 may be surrounded by a cryostat and thermal insulation (not shown).

The MRI apparatus 1 also includes gradient coils 6 embedded inside the structure of the MRI apparatus 1 and which are capable of generating magnetic field gradients inside the tunnel in three dimensions for the purpose of spatially encoding the response. The gradient coils 6 are typically resistive electromagnets with a complex amplification system for controlling the field gradient. Fig. 1 illustrates a z-dimension gradient coil 6a and a x-dimension gradient coil 6b, the MRI apparatus 1 also including a y-dimension gradient coil which is not shown.

The MRI apparatus also includes a RF transmission coil 7 and an RF receiver coil 8 for transmission and reception of RF (radio frequency) pulses.

The control system 10 of the MRI apparatus 1 is shown in Fig. 2 and may be implemented by a computer apparatus running appropriate software. The control system 10 may be a dedicated system and is disposed separately from the MRI apparatus 1, typically in another room. The control system 10 includes a user interface 15 allowing an operator to control the control system 10.

The control system 10 is connected to the MRI apparatus 1 by cables 11. To operate the MRI apparatus 1, the control system 10 generates drive signals for the main magnet 5, the gradient coils 6 and the RF transmission coil 7 and supplies them through the cables 11. RF signals received by the RF receiver coil 8 are supplied back to the control system 10 through one of the cables 11 where they are analysed by a data acquisition unit 12 to derive image data 13 which may be stored in memory and/or displayed on a display 14.

The control system 10 operates the MRI apparatus 1 to provide imaging in a conventional manner, based on the physical phenomenon of nuclear magnetic resonance. Generally the drive signals supplied to the gradient coils 6 and the RF transmission coil 7 consist of a sequence of pulses in accordance with a given drive scheme. Many different drive schemes are known and may be applied, the drive schemes providing images with different physical contrast mechanisms and

with different imaging times.

In any drive scheme, a pulse sequence having a given form, is applied repeatedly in successive periods. The length of the period is termed the repetition time (TR). Between each period, the tissue being imaged substantially returns to its original state (actually the residual magnetisation of the tissue will change over subsequent repetitions, especially over initial repetitions, but the change is small compared to the total magnetisation) so that the RF response within a period is substantially generated by the pulse sequence in the same period (ideally, although in practice a small degree of leakage between the pulses of different repetitions might be accepted to speed up the image acquisition).

Although the form of the pulse sequence in each period is the same, the actual pulses may differ in each period, for example to generate different gradient fields providing spatial encoding of different regions of the patient. For example, in a 2D Fourier Transform spin echo scheme in each successive period a phase-encoding gradient is incremented to select a different slice. Typically but without limitation, each period is associated with 1, 2 or 3 RF pulses and with one or two gradient pulses along particular directions. Typically, the control system 10 allows the number of periods to be set by the operator, for example being called the number of "scan lines" or something similar. For example, for a 2D drive scheme, each period may be associated with either one row (or column) in a resulting 2D image. Typically, each repetition time corresponds to one received burst of signal. Typically, the repetition time might be in the range from 1ms to 200ms with individual gradient pulses therewithin having lengths in the range from 0.1ms to 30ms.

In addition, before, after or during a pulse sequence there may be applied adjustment pulses to adjust the magnetisation state of the tissue.

During operation of the MRI apparatus 1, acoustic noise is generated primarily due to the pulses applied to the gradient coils 6. As the gradient coils 6 are in a region of high magnetic field generated to by the main magnet 5, when current is applied to the gradient coils 6 to adjust the magnetic field distribution across the patient 2, large Lorentz forces are generated on the gradient coils 6. The Lorentz force appears whenever current flows any direction other than directly along the magnetic field lines, and push the gradient coils 6 in a direction perpendicular to both the current flow and the magnetic field. These forces, especially since they are turned on and off with the gradient pulses, vibrate the structure of the MRI apparatus 1 generating large amounts of acoustic noise, especially within the tunnel 4 where the patient 2 is lying. To the extent that the pulse sequence is different in each period of the drive scheme, the noise will also vary in each period, limiting the ability to reduce noise by subtracting a recording of the noise in a different period.

In addition, the MRI apparatus 1 has an intercom system 20 shown in Fig. 2 and in more detail in Fig. 3. The intercom system 20 comprises a patient microphone 21, an audio processing

circuit 22 and an operator loudspeaker 23. The patient microphone 21 is located inside the tunnel 4 to pick up speech of the patient 2 and generate an audio signal 39. The patient microphone 21 may be integrated into the structure of the MRI apparatus 1 or may be separate, for example being part of a headset worn by the patient 2. However, the audio signal 39 received by the patient microphone 21 will include acoustic noise generated by the MRI apparatus 1 in addition the speech of the patient 2.

The operator loudspeaker 23 is arranged outside the MRI apparatus 1 in a location where the operator controlling the control system 10 can hear the output sound. The operator loudspeaker 23 may be integrated with the control system 10 or separate.

The audio signal 39 received by the patient microphone 21 is transmitted through the audio processing circuit 22 to the operator loudspeaker 23. In particular the audio signal 39 is supplied to a digital signal processor 25 via an ADC (analog-to-digital converter) 24 for conversion into the digital domain. The audio signal 39 is processed by the digital signal processor 25 and then supplied to the operator loudspeaker 23 via a DAC (digital-to-analog converter) 25 for conversion back into the analog domain. The to the digital signal processor 25 processes the audio signal 39 to reduce the acoustic noise in the audio signal 39 on the basis of prediction signals 26. The most important prediction signals 26 are gradient monitor signals 27 representative of the drive signals applied to the gradient coils 6 derived by a gradient monitor element 40 as will now be described. Other prediction signals 26 are also used and will be described later.

In one embodiment, the gradient monitor element 40 may be a set of pickup circuits 41 as shown in Fig. 4 arranged to derive the gradient monitor signals 27 from the one of the cables 11 supplying the drive signals from the control system 10 to the MRI apparatus 1. In this manner, the gradient monitor signals 27 may be detected externally without accessing the control system 10, allowing the noise reduction to be implemented easily on an existing MRI apparatus 1.

The pickup circuit 41 comprises a coil 42 held outside the cable 11, for example with a mechanical fixing. The coil 42 is arranged in series with a resistor 43 and in parallel with a capacitor 44 to provide a resonant circuit which picks up the derivative of the drive signal supplied along one of the cores 45 of the cable 11, as the gradient monitor signal 27 which is supplied to the digital signal processor 25 through an ADC 46 for conversion into the digital domain. The capacitor 44 is not essential. Often, sufficient capacitance is provided by the connecting cable. Similarly the resistor 43 is not essential and may be avoided, for example, if there is sufficient clamping elsewhere (e.g. losses in the coil 42). The resistor 43 may alternatively be connected in parallel with the coil 42.

Three pickup circuits 41 are arranged around the cable 11 to pick up the three drive signals supplied to the three gradient coils 6 along separate cores 45 in the cable 11. Ideally the pickup circuits 41 are each aligned with a respective core 45. However, there is leakage such that each

pickup circuit 41 is responsive to a different linear combination of the three drive signals in which the closest core predominates. For this reason, even if the pickup circuits 41 are not aligned with the cores 45, there are still derived three separate gradient monitor signals 27 which are effective for noise reduction.

In an alternative embodiment, the gradient monitor element 40 may be arranged to derive the gradient monitor signals 27 from the control system 10. This approach has the benefit of providing better quality but requires access to the control system 10 which may not be available with an existing MRI apparatus 1.

The digital signal processing circuit 25 is arranged as shown in Fig. 5 which illustrates the processing performed thereby to reduce the acoustic noise in the audio signal 39. The digital signal processing circuit 25 includes the following elements which may be implemented in hardware or by a computer apparatus running an appropriate program. The digital signal processing circuit 25 may have a real-time processing implementation or a batch processing implementation.

The audio signal 39 is supplied to a microphone filter 50 having a filter response which flattens the frequency response of the patient microphone 21 in a conventional manner.

After filtering by the microphone filter 50, the audio signal is supplied to a audio pre-emphasis filter 51 having a filter response for reducing high frequencies to reduce high frequency noise. The microphone filter 50 and audio pre-emphasis filter 51 may be arranged in the opposite order or combined in a single filter.

In general, the audio signal can contain speech of the patient 2 and acoustic noise. After filtering by the audio pre-emphasis filter 51, the audio signal is supplied to a subtractor 52 which subtracts a predicted noise signal 53 from the audio signal 39. As the predicted noise signal 53 is a prediction of the actual acoustic noise, the audio signal at the output of the subtractor 52 contains the speech of the patient and any residual noise to the extent that the prediction is inaccurate.

The audio signal output from the subtractor 52 is supplied to a gain control element 54 which adjusts the gain of the audio signal as described further below.

The audio signal output from the gain control element 54 is supplied to an inverse pre-emphasis filter 55 having a filter response which is the inverse of the filter response of the pre-emphasis filter 51. The audio signal filtered by the inverse pre-emphasis filter 55 is the output signal 70 output from the digital signal processor 25.

The predicted noise signal 53 is predicted from the prediction signals 26 by the following elements.

The prediction signals 26 are supplied to a prediction pre-emphasis filter 56 having the same filter response as the audio pre-emphasis filter 51.

After filtering by the prediction pre-emphasis filter 56, the prediction signals are supplied to a convolution unit 57 which is arranged to convolve the prediction signals with a convolution

kernel 58 to generate the predicted noise signal 53.

The convolution kernel 58 is generated by a derivation unit 59 so that the predicted noise signal 53 is a good prediction of the actual acoustic noise. This is achieved by a linear prediction approach as is known in a general sense, although there are some particular adaptations to the present situation.

The basic linear prediction approach is to assume that there is a set of linear filters for the prediction signals whose outputs can be added together to estimate the acoustic noise. The convolution kernel represents the filters, in particular the filter characteristics. Then the convolution kernel can be derived from samples of the audio signal and samples of the prediction signals, by solving linear equations representing the samples of the audio signal as the samples of the prediction signals. A time-series of samples over a predetermined period of time is used. Thus the derivation unit 59 is supplied with the audio signal after filtering by the audio pre-emphasis filter 51 and with the prediction signals after filtering by the prediction pre-emphasis filter 56.

The specific mathematical technique implemented in the derivation unit 59 is as follows. The following variables are used:

x_{ti} is the i^{th} prediction signal at time t ;

y_t is the audio signal at time t ; and

w_t is a weight function at time t , initially having a value 1 at all times and hence no effect but subsequently being used to weight the samples of the audio signal on the basis of the likelihood that the audio signal do not contain speech by having a lower value at sample times likely to contain speech than at sample times likely not to contain speech.

Cross-correlations between the samples of the audio signal and the samples of the prediction signals are accumulated over the entire time-series in accordance with the following equations for a limited range of the integer τ :

$$X_{\tau 1, \tau 2, i, j} = \sum_t x_{t-\tau 1, i} \cdot x_{t-\tau 2, j} \cdot w_{t-\tau 1} \cdot w_{t-\tau 2}$$

$$Y_{\tau, i} = \sum_t y_t \cdot x_{t-\tau, j} \cdot w_t \cdot w_{t-\tau}$$

A suitable range for $\tau, \tau 1, \tau 2$ is $\tau_{\min} \leq \tau \leq \tau_{\max}$, where $\tau_{\min}$ is typically less than or equal to zero, and $\tau_{\max}$ is greater than zero. $\tau_{\min}$ is best set by experiment, but will be proportional to reasonable measures of the length of the impulse response of the patient microphone 21 and the audio pre-emphasis filter 51. $\tau_{\max}$ is also best set by experiment, but will be typically several times the mechanical ring-down time of the system generating the noise, ie the MRI apparatus 1.

The following linear equations are derived and solved for the convolution kernel C_{sj} :

-9-

$$Y_{\tau,i} = X_{\tau,\delta,i,j}^* \cdot C_{\delta,j}$$

where

$$X_{\tau,\delta,i,j}^* = X_{\tau,\delta,i,j} + M_{\tau,\delta,i,j}$$

and $M_{\tau,\delta,i,j}$ is a regularisation term. Using conventional techniques an approximate or compromise but not exact solution is derived so as to yield a near-optimal prediction of the noise. It is advantageous to accept such a near-optimal prediction of the noise, rather than the optimal prediction, because that allows the implementation of linear regularisation by inclusion of the regularisation term $M_{\tau,\delta,i,j}$. As known in general, such linear regularisation minimises the values in the convolution kernel $C_{\delta,j}$ by seeking a solution which minimises the combination of (a) the error between the prediction and the samples of the audio signal and (b) the regularisation term $M_{\tau,\delta,i,j}$. This leads to a convolution kernel that is more reproducible, and a resulting noise estimate that is less sensitive to tiny variations in the inputs.

The regularisation term $M_{\tau,\delta,i,j}$ is non-negative definite (i.e. all the eigenvalues are $\geq$ zero) and may be expressed as a two-index matrix by lumping τ and i together into a single index, and lumping s and j together into a single index. One suitable choice for the regularisation term $M_{\tau,\delta,i,j}$ is $\lambda_{\tau,i}$ when $\tau = \delta$ and $i = j$ and zero otherwise, where $\lambda_{\tau,i} \geq 0$. Typically, one would chose $\lambda_{\tau,i}$ so that values are largest near $\tau_{\min}$ and $\tau_{\max}$ and smallest near the middle. Typically, one could choose values for $\lambda_{\tau,i}$ so that the average of $\lambda_{\tau,i}$ over τ is 0.00001 to 0.1 times the median of the absolute value of (or the RMS of) $M_{\tau,\delta,i,j}$ over τ and δ .

The convolution unit 57 employs the convolution kernel $C_{\delta,j}$ to predict noise N_t at time t in accordance with the equation:

$$N_t = \sum_i \sum_{\delta=\tau_{\min}}^{\tau_{\max}} x_{t-\delta,i} \cdot C_{\delta,i}$$

As a result, the subtractor 52 effectively calculates the noise reduced signal y'_t from the audio signal y_t in accordance with the equation:

$$y'_t = y_t - N_t$$

The above mathematical technique implemented in the derivation unit 59 is performed iteratively using the weight function w_t to weight the samples of the audio signal on the basis of the likelihood that the audio signal do not contain speech as follows.

The weight function w_t is computed as a function of the average of the noise reduced signal y'_t over a predetermined interval, for example 100ms, for example in accordance with the equation:

-10-

$$w_t = \frac{\text{median}(|y_t^r|)}{\text{median}(|y_t^r|) + \text{RMS}(y_t^r)}$$

Such an equation causes the regions which are quiet and hence likely not to contain speech to be weighted higher and hence treated as more important in the process of determining the convolution kernel $C_{\delta,j}$, and vice versa. In this equation, the numerator and the left side of the denominator are taken over a short window (typically 20ms to 500ms, preferably 50ms to 250ms) and the right side of the denominator is taken over a long window (typically longer than 200ms, preferably 300 to 3000ms). Alternatively, in off-line batch mode, the length of the longer window is the length of the entire utterance. Normally, in batch mode, the windows would be approximately centered on time t . For real-time operation, the windows would trail behind time t , using data from the recent past. And, finally, the exponents gamma should be outside the absolute value, not inside. It would be also nearly equivalent to use

$$w_t = \left(\frac{\text{median}(|y_t^r|)}{\text{median}(|y_t^r|) + \text{RMS}(y_t^r)} \right)^{\text{gamma}}$$

In either case, gamma is set by experiment, and would typically be greater than 0.5, preferably in the range 1 to 10.

Other weight functions w_t could be used including ones which take a binary value 0 or 1 on the basis of a hard decision. However it is a particular advantage that a robust decision on the likelihood that the audio signal contains speech is not necessary when using the gradient monitor signals 27 as prediction signals 26. Indeed the weight function w_t may optionally not be used at all but is useful if one of the prediction signals 26 is an audio signal from a microphone that might be correlated with speech if speech is present.

Then, an adjustment convolution kernel $C'_{\delta,j}$ is calculated using the noise reduced signal y_t^r , rather than the audio signal y_t . Finally, the sum of the previous iteration of the convolution kernel $C_{\delta,j}$ adjustment convolution kernel $C'_{\delta,j}$ is calculated to derive a new convolution kernel $C_{\delta,j}^T$ which replaces the previous iteration of the convolution kernel $C_{\delta,j}$.

Further iteration is possible.

Thus, reduction of acoustic noise is achieved using the gradient monitor signals 27 as prediction signals 26 to predict acoustic noise. The technique is capable of providing effective noise reduction because the gradient monitor signals 27 are very effective predictors. This is because acoustic noise is generated primarily due to the pulses applied to the gradient coils 6 as described above.

A further advantage is that the use of the gradient monitor signals 27 is not dependent on

an accurate classification of the audio signal as containing speech or not, because the gradient monitor signals 27 are uncorrelated with the speech of the patient 2 so there is no tendency to cancel the speech.

Optionally the noise reduction may be improved further by additionally using further prediction signals 26 including any or all of the following.

One possible prediction signal 26 is a delayed audio signal 28 derived by delaying the audio signal in a delay element 31 having a predetermined delay. Although only a single delay element 31 is illustrated, plural delay elements 31 having different delays could be used to provide plural delayed audio signals 28.

Another possible prediction signal 26 is derived from an additional microphone 32. Although a single additional microphone 32 is illustrated, plural auxiliary microphones 32 could be used. The additional microphone 29 is disposed inside the MRI apparatus 1 but further from the mouth of the patient 2 than the patient microphone 21, or even outside the MRI apparatus 1. The prediction signal 26 is a delayed auxiliary audio signal 29 derived by delaying an auxiliary audio signal supplied from the additional microphone 32 via an ADC 33 in a delay element 34 having a predetermined delay. Although only a single delay element 34 is illustrated, plural delay elements 34 having different delays could be used to provide plural delayed auxiliary audio signals 29.

Advantageously, the predetermined delays of the delay elements 31 and 34 are sufficiently long to avoid the delayed audio signal 28 and the delayed auxiliary audio signal 29 being correlated with speech if present in the audio signals received by the patient microphone 21 or the additional microphone 32, for example at least 5ms, preferably at least 15ms. Otherwise to avoid the risk that the noise prediction might also predict the speech from these signals, the derivation unit 59 would need to classify the audio signal into speech and non-speech and only derive the convolution kernel 59 from the non-speech but this is undesirable due to difficulties with performing accurate classification.

Conversely, typically the predetermined delays of the delay elements 31 and 34 are sufficiently short to provide efficient prediction, for example being at most 100ms, preferably at most 50ms.

Advantageously, the predetermined delays of the delay elements 31 and 34 are equal to, or an integer multiple (the integer not being 0 or 1) of, the repetition time of the drive signals applied to the gradient coils 6, in accordance with the drive scheme used as discussed above. This is advantageous because the drive signals in each period of the pulse sequence are very similar, although not identical. Therefore the drive signals in other periods are effective predictors of the noise. In this case, the derivation unit 59 may utilise a region of time-delays that includes the period of the pulse sequence and at least one sample on each side to allow for mismatches in the frequency responses of the various microphones. The width of this region is set by experiment, and depends

on the microphone characteristics and placement.

Another possible prediction signal 26 is an advanced audio signal from the patient microphone 21 or the additional microphone 32.

Another possible prediction signal 26 is an RF monitor signal 30 which is representative of the RF power produced by the MRI apparatus 1. The switching of the RF field can be unintentionally detected by various circuit elements including the patient microphone 21 or other elements such as amplifiers, yielding a clicking noise. The RF monitor signal 30 is useful because it allows prediction of this clicking noise.

The RF monitor signal 30 is derived by a RF monitor circuit 35 as shown in Fig. 6. The RF monitor circuit 35 includes a coil antenna 60 disposed inside or near the tunnel 4 of the RF apparatus to receive the RF field. The coil antenna 60 is connected to an amplifier 61 (which may be an operational amplifier) to amplify the received signal and supply it to a detector 62 which detects the power of the detected signal and thereby derive the RF monitor signal 30. The output of the detector 62 is supplied through a low-pass filter 63, for example having a cut-off frequency of 100kHz and then to the digital signal processor 25 through an ADC 36 for conversion into the digital domain.

As an alternative, the RF monitor signal 30 could be derived from monitoring of the drive signals applied to the RF transmission coil 7 or from the control system 10 in a similar manner to the gradient monitor signals 27.

The gain control implemented by the gain control element 54. The gain of the gain control element 54 is adjusted in a conventional manner in response to an input device such as a simple knob, thereby allowing adjustment by the operator. In addition, the gain is automatically adjusted as follows.

The gain control element 54 determines the operational state of the MRI apparatus 1, in particular whether the MRI apparatus 1 is in an operational state of generating an MRI image using the gradient coils 6. In practice this operational state typically lasts for the entire period of a series of pulse sequences used to generate one or a series of MRI images, so being a large multiple of the repetition time. This operational state may be detected in a number of ways.

One approach is to determine this operational state by detecting the presence of the gradient monitor signals 27. This is a convenient approach because the gradient monitor signals are available to the audio processing circuit 22.

Another approach is to monitor a signal supplied from the control system 10 indicating the operational state.

The gain control element 54 automatically adjusts the gain in response to the determined operational state of the MRI apparatus 1. In particular the gain control element 54 applies a higher gain when the MRI apparatus is in the operational state of generating an MRI image than when the

-13-

MRI apparatus is not in that operational state. One alternative is to have a predetermined offset between the two states so that the change in gain when the operational state changes is by the magnitude of the offset. In this case, the change of gain in response to the input device affects the gain in both operational states. Another alternative is for the gain in each operational state to be independently adjustable in response to the input device, although lower in the operational state of generating an MRI image

Such automatic adjustment therefore provides the advantage that the gain is at a lower level when in the noisy operational state of generating an MRI image, the lower gain reducing the annoyance of the acoustic noise to the operator, and is at a higher level when in the quiet state of not generating an MRI image, the higher level allowing any speech in the transmitted audio signal to be more easily heard. Otherwise, the tendency of the operator to manually reduce the gain when in the noisy operational state of generating an MRI image can result in problems of hearing the patient 2 when in the quiet state of not generating an MRI image.

The microphone filter 50 and the pre-emphasis filter 51 are conventional but for completeness their design will now be discussed.

The design of the microphone filter 50 is ultimately determined by experiments, but a good initial estimate may be obtained as follows. First the frequency response $s(\omega)$ is measured. Then the maximum desired gain F of the microphone filter 50 is estimated, typically being 3-30. Next there is computed

$S_{\max} = \max |s(\omega)|$. Then a filter gain $g(\omega)$ that will approximately flatten the frequency response over the region where $s(\omega)$ is large enough to be useful (ie where $|s(\omega)| \cdot F > 1$) is calculated by the equation:

$$g(\omega) = \frac{1}{\left(1 + \left(\frac{|s(\omega)|}{S_{\max}} \cdot F\right)^2\right)^{0.5} \cdot \left(1 + \left(\frac{|s(\omega)|}{S_{\max}} \cdot F\right)^{-2}\right)^{0.5}}$$

For a batch processing implementation, it is quite reasonable to filter via $S(t) =$

$F^{-1} [g(\omega) \cdot F[s(t)]]$ which is a standard acausal (time-symmetric) implementation of a filter via Fourier Transforms. For real-time processing, one would use well known techniques to construct a causal IIR or FIR filter that has a frequency response whose magnitude approximates $g(\omega)$.

The design of the pre-emphasis filter 51 is ultimately determined by experiments, but a good initial estimate may be obtained as follows. First, the typical speech power spectrum $P(\omega)$ after filtering by the microphone filter 50 is measured. Next the typical power spectrum $M(\omega)$ of the MRI apparatus at the same point is measured. Then a first-order IIR filter is constructed using

-14-

standard techniques to have a frequency response $E(\omega)$ given by the equation:

$$\int_{\omega_1}^{\omega_2} (P(\omega)M(\omega))^{\frac{1}{2}} |E(\omega)|^2 d\omega = \int_{\omega_3}^{\omega_4} (P(\omega)M(\omega))^{\frac{1}{2}} |E(\omega)|^2 d\omega$$

where $\omega_1=2\pi \cdot 50\text{Hz}$, $\omega_2=2\pi \cdot 500\text{Hz}$, $\omega_3=2\pi \cdot 1000\text{Hz}$, $\omega_4=2\pi \cdot 2500\text{Hz}$. This choice of ω_1 to ω_4 arranges for a compromise between flattening the speech spectrum and flattening the noise spectrum.

There is described above noise reduction in an intercom system 20 transmitting an audio signal 39 from a patient microphone 21 to an operator loudspeaker 23. Additionally, the intercom system 20 will typically also be arranged to transmit an audio signal from an operator microphone arranged outside the MRI apparatus 1 in a location to pick up the speech of the operator to a patient loudspeaker located inside the tunnel 4 in a location where the patient 2 can hear the output sound. In this case, the acoustic noise of the MRI apparatus 1 will be heard by the patient 2 and will mask any sound output from the patient loudspeaker. The acoustic noise can be reduced but not removed by the patient 2 wearing earphones which cover the ears and absorb ambient sound, the patient loudspeaker in this case being arranged inside the earphones. The residual acoustic noise can be reduced by using an audio processing circuit similar to that described above to predict the acoustic noise using equivalent prediction signals and subtract the predicted acoustic noise from the signal provided to the patient loudspeaker, effectively cancelling the residual acoustic noise penetrating the earphones. In this case, an additional microphone is arranged inside the earphones to detect the acoustic noise.

More generally, the noise reduction technique described above could be generalised to provide noise reduction in any system having a noise-generating element which generates periodic noise, for example an aircraft propellor. In this general case, the gradient monitor signals 27 are generalised to prediction signals which are representative of the drive signals applied to the noise-generating element or more generally still representative of the mechanical state of the noise-generating element, for example the angular position of an aircraft propellor. The acoustic noise is predicted from the prediction signals in the same manner as discussed above. The predicted acoustic noise is subtracted from an audio signal applied to a loudspeaker to effectively cancel the noise in the same manner as described above.

Claims

1. A method of reducing acoustic noise generated by a MRI apparatus in an audio signal transmitted by an intercom system from a patient microphone disposed inside the MRI apparatus to an operator loudspeaker disposed outside the MRI apparatus, the method comprising:
 - deriving gradient monitor signals representative of the drive signals applied to the gradient coils of the MRI apparatus;
 - predicting the acoustic noise generated by the MRI apparatus from prediction signals including at least the gradient monitor signals; and
 - subtracting the predicted acoustic noise from the audio signal transmitted by the intercom system.
2. A method according to claim 1, wherein the prediction signals further include the audio signals transmitted by the intercom system, delayed by a one or more predetermined delays.
3. A method according to claim 2, wherein the one or more predetermined delays in respect of the audio signals transmitted by the intercom system are at least 5ms, preferably at least 15ms.
4. A method according to claim 2 or 3, wherein the one or more predetermined delays in respect of the audio signals transmitted by the intercom system are at most 100ms, preferably at most 50ms.
5. A method according to any one of claims 2 to 4, wherein the one or more predetermined delays in respect of the audio signals transmitted by the intercom system are equal to or a multiple of the repetition time of the drive signals.
6. A method according to claim 5, wherein the prediction signals further include auxiliary audio signals output by at least one additional microphone disposed inside or adjacent the MRI apparatus, delayed by a one or more predetermined delays.
7. A method according to claim 6, wherein the one or more predetermined delays in respect of the auxiliary audio signals are at least 5ms.
8. A method according to claim 6 or 7, wherein the one or more predetermined delays in respect of the auxiliary audio signals are at most 100ms, preferably at most 50ms.

-16-

9. A method according to any one of claims 6 to 8, wherein the one or more predetermined delays in respect of the auxiliary audio signals are equal to or a multiple of the repetition time of the drive signals.
10. A method according to any one of the preceding claims, wherein the prediction signals further include RF monitor signals representative of the RF power produced by the MRI apparatus.
11. A method according to any one of the preceding claims, wherein the gradient monitor signals are derived by a pick-up circuit arranged outside a cable supplying the drive signals to the gradient coils of the MRI apparatus.
12. A method according to any one of claims 1 to 10, wherein the gradient monitor signals are derived from a control system which generates the drive signals.
13. A method according to any one of the preceding claims, wherein the step of predicting the acoustic noise generated by the MRI apparatus comprises convolving the prediction signals with a convolution kernel, the method further comprising deriving the convolution kernel from samples of the audio signal and samples of the prediction signals.
14. A method according to claim 13, wherein the step of deriving the convolution kernel from samples of the audio signal and samples of the prediction signals comprises solving equations representing the samples of the audio signal as the samples of the prediction signals convolved by the convolution kernel, to derive the convolution kernel.
15. A method according to claim 14, wherein said equations include regularisation terms for regularising the solution for the convolution kernel.
16. A method according to claim 14 or 15, wherein said step of solving equations to derive the convolution kernel is iterated.
17. A method according to any one of claims 14 to 16, wherein the step of deriving the convolution kernel from samples of the audio signal and samples of the prediction signals comprises weighting the samples of the audio signal on the basis of the likelihood that the audio signal do not contain speech, said equations representing the weighted samples of the audio signal as the samples of the prediction signals convolved by the convolution kernel.

-17-

18. An intercom system for transmitting sound from a patient microphone disposed inside the MRI apparatus to an operator loudspeaker disposed outside the MRI apparatus, the intercom system comprising:

gradient monitor means for deriving gradient monitor signals representative of the drive signals applied to the gradient coils of the MRI apparatus;

prediction means for predicting the acoustic noise generated by the MRI apparatus from prediction signals including at least the gradient monitor signals; and

noise reduction means for subtracting the predicted acoustic noise from the audio signal transmitted by the intercom system.

19. An intercom system according to claim 18, further comprising delay means for delaying the audio signals transmitted by the intercom system by a one or more predetermined delays to derive delayed audio signals, said prediction signals used by said prediction means further including the delayed audio signals.

20. An intercom system according to claim 19, wherein the one or more predetermined delays in respect of the audio signals transmitted by the intercom system are at least 5ms, preferably at least 15ms.

21. An intercom system according to claim 19 or 20, wherein the one or more predetermined delays in respect of the audio signals transmitted by the intercom system are at most 100ms, preferably at most 50ms.

22. An intercom system according to claims 19 to 21, wherein the one or more predetermined delays in respect of the audio signals transmitted by the intercom system are equal to or a multiple of the repetition time of the drive signals to the gradient coils of the MRI apparatus.

23. An intercom system according to any one of claims 18 to 22, further comprising:

at least one additional microphone disposed inside or adjacent the MRI apparatus to output auxiliary audio signals; and

delay means for delaying the auxiliary audio signals by a one or more predetermined delays to derive delayed auxiliary audio signals, said prediction signals used by said prediction means further including the delayed auxiliary audio signals.

24. An intercom system according to claim 23, wherein the one or more predetermined delays

in respect of the delayed auxiliary audio signals are at least 5ms, preferably at least 15ms.

25. An intercom system according to claim 23 or 24, wherein the one or more predetermined delays in respect of the delayed auxiliary audio signals are at most 100ms, preferably at most 50ms.

26. An intercom system according to any one of claims 23 to 25, wherein the one or more predetermined delays in respect of the delayed auxiliary audio signals are equal to or a multiple of the repetition time of the drive signals to the gradient coils of the MRI apparatus.

27. An intercom system according to any one of claims 18 to 26, further comprising RF monitor means for deriving RF monitor signals representative of the RF power produced by the MRI apparatus, said prediction signals used by said prediction means further including the RF monitor signals.

28. An intercom system according to any one of claims 18 to 27, wherein the gradient monitor means comprises a pick-up circuit arranged outside a cable supplying the drive signals to the gradient coils of the MRI apparatus.

29. An intercom system according to any one of claims 18 to 27, wherein the gradient monitor means comprises means for extracting the gradient monitor signals from a control system which generates the drive signals.

30. An intercom system according to any one of claims 18 to 29, wherein
the prediction means is arranged to predict the acoustic noise generated by the MRI apparatus by convolving the prediction signals with a convolution kernel,
the intercom system further comprising derivation means for deriving the convolution kernel from samples of the audio signal and samples of the prediction signals.

31. An intercom system according to claim 30, wherein the derivation means is arranged to solve equations representing the samples of the audio signal as the samples of the prediction signals convolved by the convolution kernel, to derive the convolution kernel.

32. An intercom system according to claim 31, wherein said equations include regularisation terms for regularising the solution for the convolution kernel.

33. An intercom system according to claim 30 or 31, wherein said derivation means is arranged to solve said equations iteratively.

34. An intercom system according to any one of claims 30 to 33, wherein said derivation means is arranged to weight the samples of the audio signal on the basis of the likelihood that the audio signal do not contain speech, said equations representing the weighted samples of the audio signal as the samples of the prediction signals convolved by the convolution kernel.

35. A method of automatically controlling the gain of an intercom system of an MRI apparatus transmitting an audio signal from a patient microphone disposed inside the MRI apparatus to an operator loudspeaker disposed outside the MRI apparatus, the method comprising:

determining whether or not the MRI apparatus is in an operational state of generating an MRI image using the gradient coils of the MRI apparatus; and

controlling the gain of the intercom system to apply a lower gain when it is determined that the MRI apparatus is in said operational state than when it is determined that the MRI apparatus is not in said operational state.

36. An intercom system for an MRI apparatus arranged to transmit an audio signal from a patient microphone for disposal inside the MRI apparatus to an operator loudspeaker for disposal outside the MRI apparatus,

the intercom system having a gain control circuit for adjusting the gain of the audio signal, the gain control circuit being responsive to whether or not the MRI apparatus is in an operational state of generating an MRI image using the gradient coils of the MRI apparatus, to apply a lower gain when the MRI apparatus is in said operational state than when the MRI apparatus is in said operational state.

1/3

Fig.1.

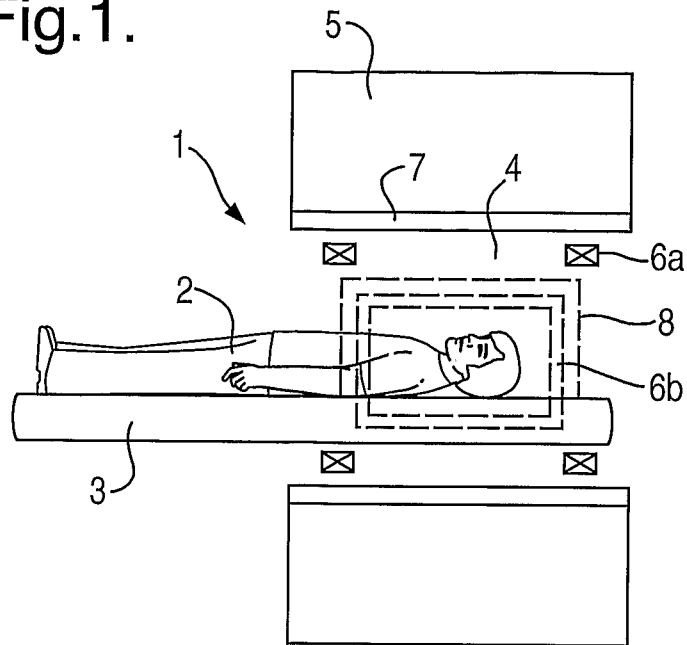
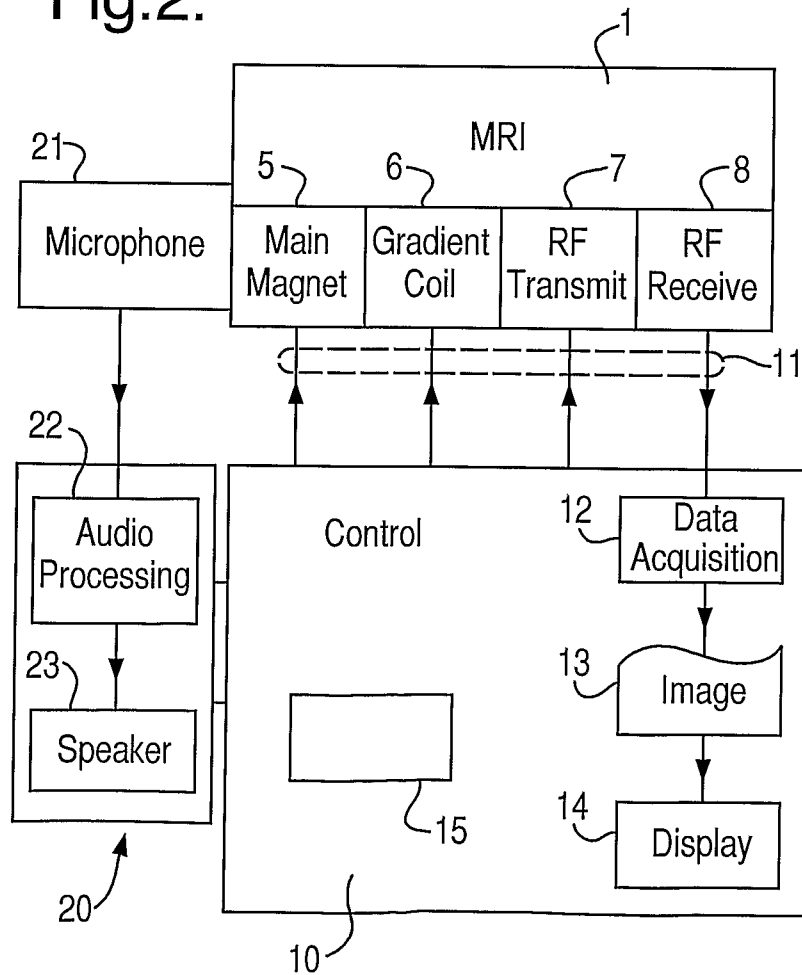


Fig.2.



2/3

Fig.3.

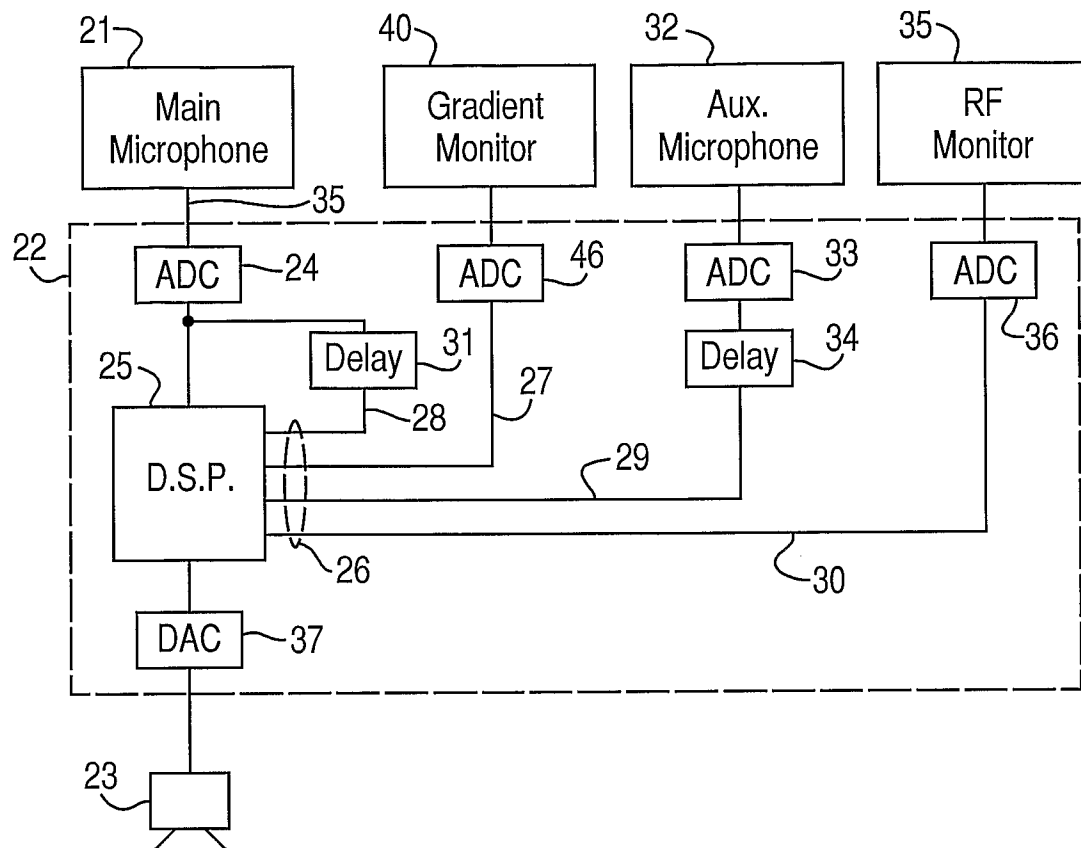
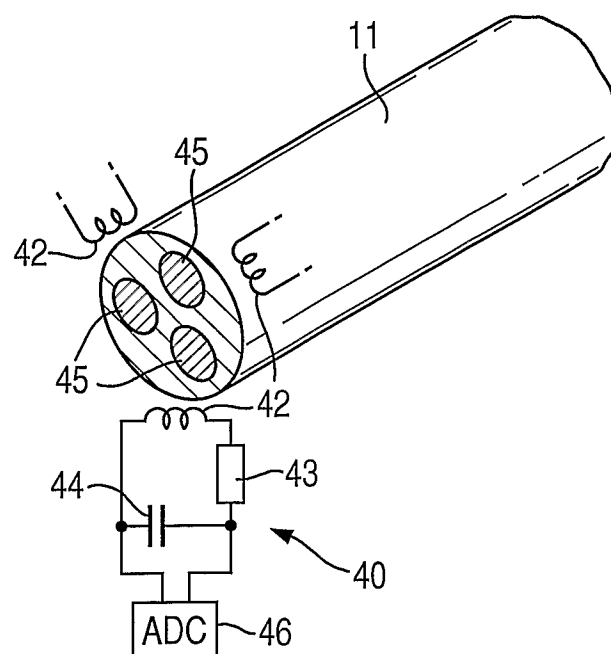


Fig.4.



3/3

Fig.5.

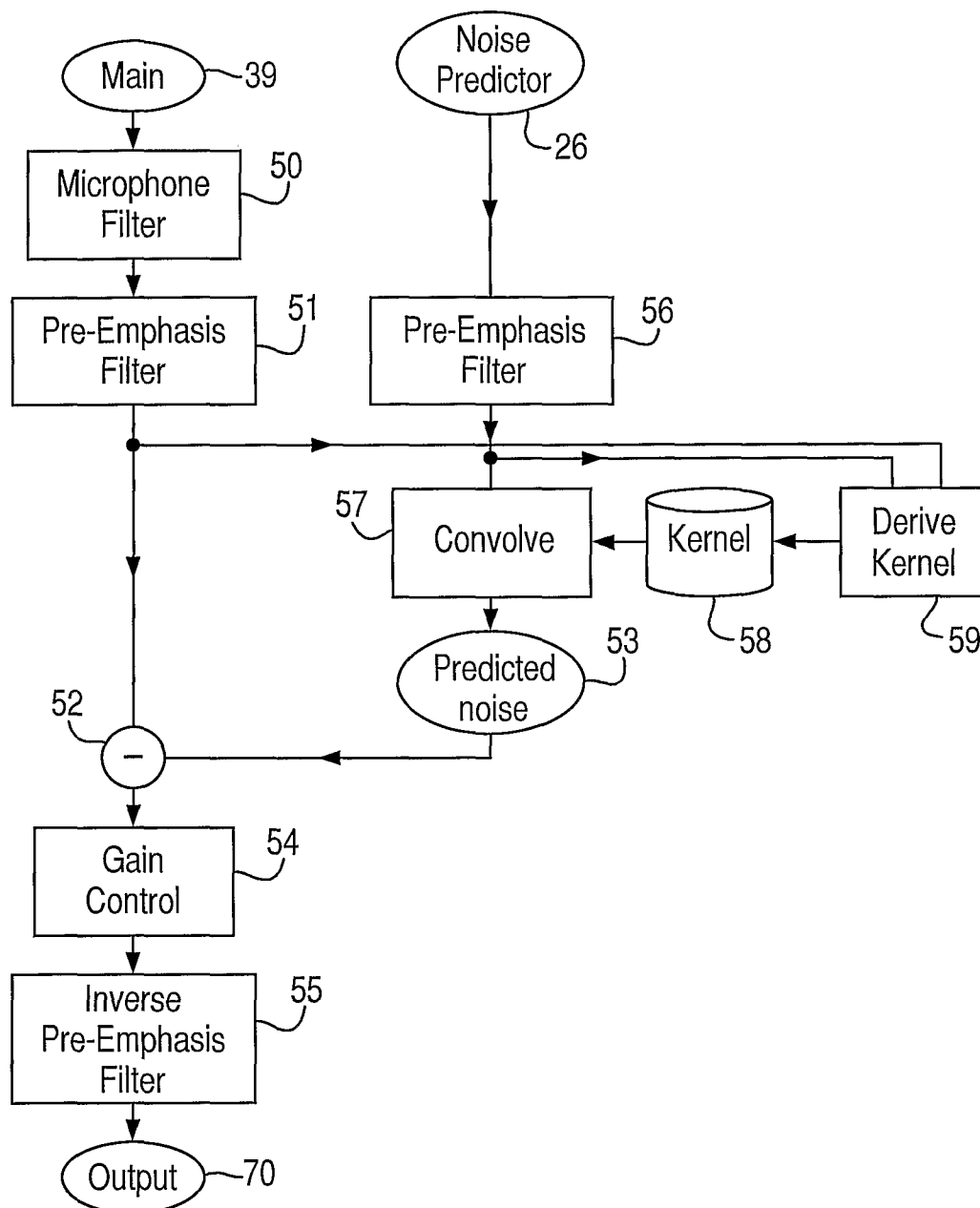
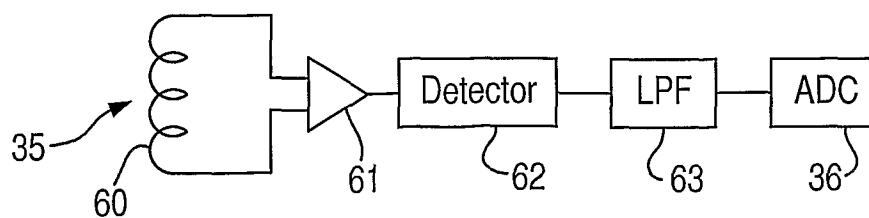


Fig.6.



INTERNATIONAL SEARCH REPORT

International application No
PCT/GB2009/002634

A. CLASSIFICATION OF SUBJECT MATTER
INV. G01R33/28 G10L21/02 H04B1/12 H04B15/00

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

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

G01R G10L H04B

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

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

EPO-Internal, WPI Data

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	US 4 696 030 A (EGOZI AVIHAY [IL]) 22 September 1987 (1987-09-22) cited in the application column 3 - column 4	1-34
X	JP 2000 333931 A (YOKOGAWA MEDICAL SYST) 5 December 2000 (2000-12-05) abstract; claims 1,2; figure 1 paragraphs [0011] - [0017] -/--	1,11-18, 28-34

☒ Further documents are listed in the continuation of Box C.

☒ See patent family annex.

* Special categories of cited documents :

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

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

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

"O" document referring to an oral disclosure, use, exhibition or other means

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

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

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

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

"&" document member of the same patent family

Date of the actual completion of the international search

15 January 2010

Date of mailing of the international search report

22/04/2010

Name and mailing address of the ISA/

European Patent Office, P.B. 5818 Patentlaan 2
NL - 2280 HV Rijswijk
Tel. (+31-70) 340-2040,
Fax: (+31-70) 340-3016

Authorized officer

Raguin, Guy

INTERNATIONAL SEARCH REPORT

International application No
PCT/GB2009/002634

C(Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	HEDEEN R A ET AL: "CHARACTERIZATION AND PREDICTION OF GRADIENT ACOUSTIC NOISE IN MR IMAGERS" MAGNETIC RESONANCE IN MEDICINE, ACADEMIC PRESS, DULUTH, MN, US, vol. 37, no. 1, 1 January 1997 (1997-01-01), pages 7-10, XP000636876 ISSN: 0740-3194 abstract; figures 1,4 page 9 - page 10	1,11-18, 28-34
Y	US 6 407 548 B1 (DIETZ PETER [DE]) 18 June 2002 (2002-06-18) column 3, line 53 - column 6, line 11; figure 1	1,11-18, 28-34
Y	SIERRA C V R ET AL: "Acoustic fMRI Noise: Linear Time-Invariant System Model" IEEE TRANSACTIONS ON BIOMEDICAL ENGINEERING, IEEE SERVICE CENTER, PISCATAWAY, NJ, US, vol. 55, no. 9, 1 September 2008 (2008-09-01), pages 2115-2123, XP011246836 ISSN: 0018-9294 page 2115, column 2 - page 2116, column 1; figures 1,3,4 page 2119, column 2, paragraph 2 page 2120, column 2, paragraph 4	1,11-18, 28-34
Y	YAO ET AL: "Acoustic noise simulation and measurement of a gradient insert in a 4T MRI" APPLIED ACOUSTICS, ELSEVIER PUBLISHING, GB, vol. 66, no. 8, 1 August 2005 (2005-08-01), pages 957-973, XP005388164 ISSN: 0003-682X abstract; figures 7-9 page 959 - page 963 page 972	1,11-18, 28-34
Y	CASPER K CHEN ET AL: "Active Cancellation System of Acoustic Noise in MR Imaging" IEEE TRANSACTIONS ON BIOMEDICAL ENGINEERING, IEEE SERVICE CENTER, PISCATAWAY, NJ, US, vol. 46, no. 2, 1 February 1999 (1999-02-01), XP011006653 ISSN: 0018-9294 the whole document	2-9, 19-26

-/--

INTERNATIONAL SEARCH REPORT

International application No

PCT/GB2009/002634

C(Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	CUSACK R ET AL: "Automated post-hoc noise cancellation tool for audio recordings acquired in an MRI scanner" HUMAN BRAIN MAPPING, WILEY-LISS, NEW YORK, NY, US, vol. 24, no. 4, 1 April 2005 (2005-04-01), pages 299-304, XP007911113 ISSN: 1065-9471 cited in the application the whole document	2-9, 19-26
Y	ROESCHMANN P: "HUMAN AUDITORY SYSTEM RESPONSE TO PULSED RADIOFREQUENCY ENERGY IN RF COILS FOR MAGNETIC RESONANCE AT 2.4 TO 170 MHZ" MAGNETIC RESONANCE IN MEDICINE, ACADEMIC PRESS, DULUTH, MN, US, vol. 21, no. 2, 1 October 1991 (1991-10-01), pages 197-215, XP000235326 ISSN: 0740-3194 the whole document	10,27
A	US 6 963 649 B2 (VAUDREY MICHAEL A [US] ET AL) 8 November 2005 (2005-11-08) cited in the application the whole document	1-34

INTERNATIONAL SEARCH REPORT

International application No.
PCT/GB2009/002634

Box No. II Observations where certain claims were found unsearchable (Continuation of item 2 of first sheet)

This international search report has not been established in respect of certain claims under Article 17(2)(a) for the following reasons:

1. ☐ Claims Nos.:
because they relate to subject matter not required to be searched by this Authority, namely:
2. ☐ Claims Nos.:
because they relate to parts of the international application that do not comply with the prescribed requirements to such an extent that no meaningful international search can be carried out, specifically:
3. ☐ Claims Nos.:
because they are dependent claims and are not drafted in accordance with the second and third sentences of Rule 6.4(a).

Box No. III Observations where unity of invention is lacking (Continuation of item 3 of first sheet)

This International Searching Authority found multiple inventions in this international application, as follows:

see additional sheet

1. ☐ As all required additional search fees were timely paid by the applicant, this international search report covers all searchable claims.
2. ☐ As all searchable claims could be searched without effort justifying an additional fees, this Authority did not invite payment of additional fees.
3. ☐ As only some of the required additional search fees were timely paid by the applicant, this international search report covers only those claims for which fees were paid, specifically claims Nos.:
4. ☒ No required additional search fees were timely paid by the applicant. Consequently, this international search report is restricted to the invention first mentioned in the claims; it is covered by claims Nos.:

1-34

Remark on Protest

- ☐ The additional search fees were accompanied by the applicant's protest and, where applicable, the payment of a protest fee.
- ☐ The additional search fees were accompanied by the applicant's protest but the applicable protest fee was not paid within the time limit specified in the invitation.
- ☐ No protest accompanied the payment of additional search fees.

FURTHER INFORMATION CONTINUED FROM PCT/ISA/ 210

This International Searching Authority found multiple (groups of) inventions in this international application, as follows:

1. claims: 1-34

Method and corresponding apparatus, wherein the acoustic noise generated by a MRI apparatus in an audio signal transmitted by an intercom system is reduced, wherein gradient monitor signals are used to predict the acoustic noise, after which the predicted acoustic noise is subtracted from the audio signal.

2. claims: 35,36

Method and corresponding apparatus of automatically controlling the gain of an intercom system of a MRI apparatus system.

INTERNATIONAL SEARCH REPORT

Information on patent family members

International application No

PCT/GB2009/002634

Patent document cited in search report		Publication date	Patent family member(s)	Publication date
US 4696030	A	22-09-1987	NONE	
JP 2000333931	A	05-12-2000	JP 4162329 B2	08-10-2008
US 6407548	B1	18-06-2002	DE 19903627 A1	03-08-2000
US 6963649	B2	08-11-2005	US 2006002570 A1	05-01-2006
			US 2002048377 A1	25-04-2002