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(54) Background noise estimation in a loudspeaker-room-microphone system

(57) A system and a method are disclosed in which the signal components of a plurality of non-interdependent acoustic channels of a multichannel music source are processed in a multichannel assembly for acoustic echo cancellation (AEC) so that the music components in the signal of a microphone are optimally cancelled. The microphone is located in a closed-off acoustic room, such as, for example, the passenger compartment of a motor vehicle and picks up signal components of music, background noise and any speech signals existing.

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Description

BACKGROUND

1. Field of Invention

[0001] The invention relates to an assembly for multichannel echo cancellation, particularly to an assembly for multichannel cancellation of the music signal component of a microphone signal.

2. Related Art

[0002] In rooms closed off acoustically, such as for instance the passenger compartment of a motor vehicle, it is often wanted to implement estimating the background noise and/or to process and optimize a speech signal of a passenger to enhance communication within the room. The term background noise in this context generally includes both acoustic waves acting from outside such as, for example, environmental noise or the noise of the vehicle on the move as picked up in the passenger compartment of a motor vehicle, as well as acoustic waves triggered by vibration, for instance of the passenger compartment or transmission of a motor vehicle. When this noise is unwanted it is also termed nuisance noise.

[0003] When music or speech is relayed by an electro-acoustic (audio) system in a noisy environment such as, for instance the passenger compartment of a motor vehicle, this may likewise be a nuisance to voice communication. This background noise may involve noise stemming from the wind, the engine of the motor vehicle, the tyres, a blower and other components of the motor vehicle and is thus a function of the speed, tyre/road contact and operating conditions of the motor vehicle on the move. Many motor vehicles nowadays feature entertainment systems involving high-end audio signal replication via a plurality of loudspeakers arranged in the passenger compartment of the motor vehicle.

[0004] This means that music signals now often exist simultaneously with background noise and speech signals. To implement estimating background noise and/or to process and optimize a speech signal of a vehicle occupant in the passenger compartment usually at least one microphone is arranged in the passenger compartment, the signals of which are correspondingly processed. Since it is desirable in this context to cancel the music components in the microphone signal so that ideally only the signal components of the background noise and speech signals remain, there is a need to provide an assembly to cancel the music component of a microphone signal.

SUMMARY

[0005] This is achieved by an assembly in which the signal components of a plurality of non-interdependent acoustic channels of a multichannel music source are processed in a multichannel assembly for acoustic echo cancellation (AEC) so that the music components in the signal of a microphone are optimally cancelled. In this arrangement the microphone is located in a closed-off acoustic room, such as, for example, the passenger compartment of a motor vehicle and picks up signal components of music, background noise and any speech signals existing.

BRIEF DESCRIPTION OF THE DRAWINGS

[0006] The invention can be better understood with reference to the following drawings and description. The components in the figures are not necessarily to scale, emphasis is instead placed on illustrating the principles of the invention. Moreover, in the figures, like reference numerals designate corresponding parts throughout the different views.

FIG. 1 is a diagram showing the basic principle of an adaptive filter;

FIG. 2 is a diagram showing the function principle of an LMS algorithm;

FIG. 3 is a diagram showing a structure of an LRM assembly for estimating the transfer function between the loudspeaker and microphone;

FIG. 4 is a diagram showing an assembly in accordance with the invention for two-channel echo cancellation;

FIG. 5 is a diagram showing an assembly for generating multichannel music signals;

FIG. 6 is a block circuit diagram showing an assembly of a seven-channel audio system as shown in FIG. 5 in a given room; and

FIG. 7 is a diagram showing an assembly in accordance with the invention for multichannel echo cancellation.

DETAILED DESCRIPTION

[0007] Referring now to FIG. 1 adaptive filters are used in recursive methods to approximate a wanted impulse response or the transfer function of an unknown system with sufficient accuracy, as is also termed estimating the transfer function of an unknown system. Adaptive filters are understood to be digital filters realized typically with the aid of algorithms on digital signal processors and which adapt their filter coefficients by a given algorithm to an input signal. In this arrangement an unknown system is assumed to be a linear distorting system whose transfer function is sought. To find this transfer function an adaptive system is circuited in parallel with the unknown system.

[0008] Adaptive processes have the advantage that by continually changing the filter coefficients the algorithms automatically adapt also to changing conditions of the surroundings, for example, room changes due to different passenger and luggage situations within the passenger compartment. This capability is achieved by a recursive system structure which continually optimizes the parameters. The unknown system may be, for example, the passenger compartment of a motor vehicle in which a signal (for example speech and/or music) projected by one or more loudspeakers, is filtered via the unknown transfer function of the room and picked up by a microphone in this room. The basic principle of an adaptive filter realized in a digital signal processor is shown in FIG. 1.

[0009] The arrangement of FIG. 1 comprises an "unknown system" U and an "adaptive filter" A. The unknown system U may be, for example the unknown acoustic transfer function of the passenger compartment of a motor vehicle. As shown in FIG. 1 an input signal x(n) is converted by the unknown system U, for example an acoustic transfer path into a signal d(n). In addition, the input signal x(n) is translated by the adaptive filter A into the signal y(n) .

[0010] As evident from FIG. 1 the signal d(n) distorted by the transfer function of the unknown system U serves as the desired reference signal from which the output y(n) of the adaptive filter A is deducted in thus generating an error signal e(n). Using for example the least mean square (LMS) method, the filter coefficients are set by iteration so that the error signal e(n) is minimized, resulting in y(n) approximating d(n). This achieves approximation of the unknown system U and thus also of its transfer function in maximizing extinction of the signal d(n) by the signal y(n).

[0011] In this arrangement the LMS algorithm is an algorithm for approximating the solution of the known LMS problem as often occurs, for example, in application of adaptive filters realized in digital signal processors. The algorithm is based on the so-called method of steepest descent (gradient descent method) for simple approximation of the gradient.

[0012] The algorithm works recursive in time, i.e. with every new set of data the algorithm is reactivated and the solution updated. Because of its low complexity, numerical stability and low memory requirement the LMS algorithm is often employed for adaptive filters and adaptive controls. Further methods could be, for example, recursive least squares, QR decomposition least squares, least squares lattice, zero-forcing, stochastic gradient methods, etc.

[0013] Adaptive filters include infinite impulse response (IIR) filters or finite impulse response (FIT) filters. FIR filters are characterized by having a finite impulse response and working in discrete steps in time as are usually determined by the sampling frequency of an analog signal. A FIR filter of the n^{th} order is described by the differential equation:

$$y(n) = b_0 * x(n) + b_1 * x(n-1) + b_2 * x(n-2) + \ldots + b_{N-1} * x(n-N-1)$$
$$= \sum_{i=0}^{N-1} b_i * x[n-i]$$

where y(n) is the starting value at the point in time n as computed from the sum of the N last sampled input values x(n-N-1) to x(n) weighted with the filter coefficients b_i. The transfer function to be approximated is realized as described above, for example, by modifying these filter coefficients b_i.

[0014] Unlike FIR filters, IIR filters also take into account the already computed starting values (recursive filter) and are characterized by having an infinite impulse response. But since the computed values are very small after a finite time, computation can be discontinued after a finite number of sampling values n in actual practice. The specification for the computation of an IIR filter is:

$$y(n) = \sum_{i=0}^{N-1} b_i * x(n-i) - \sum_{i=0}^{M-1} a_i * y(n-i)$$

where y(n) is the starting value at the point in time n as computed from the sum of the sampled input values x(n) weighted with the filter coefficients b_i added to the sum of the starting values y(n) weighted with the filter coefficients a_i. Specifying the filter coefficients a_i and b_i realizes in turn the wanted transfer function.

[0015] Unlike FIR filters, IIR filters may be unstable, but attain higher selectivity for the same expense in realization. In actual practice the filter that is selected is the filter which best satisfies the necessary conditions, taking into account the requirements and the complexity of the computation involved.

[0016] Referring now to FIG. 2 there is illustrated diagrammatically the sequence of a typical LMS algorithm for iterative adaptation of a FIR filter by way of example. FIG. 2 includes the reference signal x(n) already known from FIG. 1 as a first input signal for the adaptive LMS algorithm as well as, as a second input signal, the signal d(n) resulting from x(n) distorted as shown in FIG. 1 from the unknown system by the transfer function thereof.

[0017] How these two input signals are generated depends on the wanted application. As already explained above, these input signals may be sound signals converted by microphones into electrical signals, but just as well may include electrical signals, generated for example by sensors for picking up mechanical oscillations or also by tachometers, for example in means for reducing the noise in motor vehicles.

[0018] Furthermore included in FIG. 2 is the diagrammatic representation of an n^{th} order filter (depicted here in the embodiment of a FIR filter by way of example) by which a reference signal x(n) is converted into a signal y(n) at the point in time n. The filter coefficients of the adaptive filter as are actual at the point in time n as shown in FIG. 2 are identified $b_0(n)$, $b_1(n)$ ... $b_{N-1}(n)$. As evident from FIG. 2 the adaptation algorithm changes the filter parameters iteratively until the error signal or difference signal e(n) between the signal d(n), distorted by the transfer function of the unknown system, and the filtered reference signal y(n)is minimized.

[0019] The two input signals x(n) and d(n) are generally stochastic signals, in the case of acoustic echo cancellation (AEC) systems, for example, noisy detected signals, activation signals or communication signals. The quality factor of the adaptation is thus often taken as the power of the error signal e(n) or the mean squared error (MSE), where

$$MSE = E\{e^2[n]\}$$

[0020] The quality factor as expressed by the MSE can be minimized by a simple recursive algorithm, the said least mean square (LMS) algorithm.

[0021] In the least mean square method the function to be minimized is the square of the error, meaning that for a better approximation of the minimum of the error squared, simply the error itself, multiplied by a constant, is added to the approximation as last found before. The adaptive FIR filter must be selected at least as long as the relevant component of the unknown impulse response of the unknown system to be approximated so that the adaptive filter has a sufficient degree of freedom to really minimize the error signal e(n).

[0022] The filter coefficients are changed stepwise in the direction of the maximum reduction or negative gradient of the degree of error MSE, the parameter μ regulating the step size.

[0023] A typical LMS algorithm for computing the filter coefficient $b_k(n)$ of an adaptive filter, such as, for instance, a FIR filter as used by way of example in the further sequence can be described as follows:

$$b_k[n+1] = b_k[n] + 2\cdot\mu\cdot e[n]\cdot x[n-k] \text{ for } k=0,\ldots,N-1.$$

[0024] The new filter coefficients $b_k(n+1)$ correspond to the old filter coefficients $b_k(n)$ plus a correctional term which depends on the error signal e(n) (see FIG. 1) and the value x(n-k) assigned to each filter coefficient b_k. The LMS convergence parameter μ, also referred to as gain factor or step size, represents a measure for how fast and stable the filter is adapted.

[0025] The adaptive filter, in the present example a FIR filter, in application of the LMS algorithm converges to the well known Wiener filter when the following applies to the gain factor μ

$$0 < \mu < \mu_{max} = 1/[N\cdot E\{x^2[n]\}]$$

where N is the order of the FIR filter and $E\{x^2[n]\}$ represents the signal strength of the reference signal x(n). In actual practice the step size or convergence parameter μ used is often selected $\mu = \mu_{max}/10$.

[0026] It is in this way that the LMS algorithm of the adaptive LMS filter can be realized as described in the following:

1. Initialize the algorithm: Set the running variable n to n = 0 and select the starting coefficients $b_k(n=0)$ for k = 0,..., N-1 to start performance of the algorithm, whereby $b_k(0)= 0$ for k = 0,...,N-1 is a suitable selection, since on starting the algorithm e(0) = d(0). Then select the gain factor $\mu < \mu_{max}$, typically $\mu = \mu_{max}/10$

2. Enter the reference value x(n) and the signal d(n) distorted by the transfer function of the unknown system.

3. FIR filtering of the reference signal according to

$$y[n] = \sum_{k=0}^{N-1} b_k[n] \cdot x[n-k]$$

4. Determine the error: e(n) = d(n)-y(n)

5. Update the coefficients in accordance with $b_k[n+1] = b_k[n] + 2\cdot\mu\cdot e[n]\cdot x[n-k]$ for k=0, ...,N-1.

6. Prepare the next step in iteration n = n+1 and continue as per step 2.

[0027] It can readily be seen from the sequence of the LMS algorithm, as described, that a filter based on an LMS algorithm converges faster, the larger the convergence parameter μ, in other words the possible size between the steps in iteration is selected. At the same time it can be seen that also the quality factor of the achievable mean squared error (MSE) depends on this step size, i.e. the convergence parameter μ. The smaller μ is selected the less is the final deviation from the iterative approximated target value, i.e. the smaller the error signal e(n) achieved with the aid of the adaptive filter. A small error signal e(n), ideally an error signal e(n) = 0 is desired.

[0028] However, selecting a relatively small convergence parameter μ also simultaneously necessitates a larger number of iterations for approximating the wanted target value, as a result the convergence time needed by the adaptive filter increases. This is why in actual practice selecting the convergence parameter μ is always a compromise between the quality of the approximation to the target value and the speed of the adaptation.

[0029] It is apparent that with regard to the wanted achievable accuracy of the adaptation a typically small step size μ is selected. But with a small step size μ there may be the drawback that adapting the LMS algorithm cannot be adapted fast enough to the reference signal changing quickly in time of the assembly, for instance transient pulsating sound components which then cannot be reduced to the desired degree by extinction. Indeed, when the step size μ is too large in an extreme case this - as already explained above - may even result in an unwanted instability of the adaptation algorithm as a whole where signals fast-changing in time are concerned.

[0030] It is especially in applications for echo cancellation that the so-called normalized LMS algorithm (NLMS algorithm) is often employed in actual practice. The main drawback of the LMS algorithm is that, as described, it may react sensitive to changes in the size of the input signal x(n). The normalized LMS algorithm is a variant of the LMS algorithm which gets round this problem by normalizing the step size μ with the actual power of the input signal x(n), a normalized step size μ_n being equated as

$$\mu_n = \mu / \| x(n) \|^2$$

where $\|x(n)\|$ is the Euclidian vector standard of x(n).

[0031] Referring now to FIG. 3, there is illustrated, by way of example, a loudspeaker room microphone (LRM) system employing an adaptive filter for echo cancellation, it shows a loudspeaker L, a signal source S and a microphone M. Also shown in FIG. 3 is a signal x(n) and the impulse response h(n) of the transfer path between the loudspeaker L and the microphone M. In addition, FIG. 3 shows the principle structure of a signal processing path for cancellation of echo signals, this signal processing path comprising an adaptive filter $\hat{h}(n)$ and a summing block Σ_1. FIG. 3 thus illustrates how a feedback signal is generated from the signal x(n) for activating the loudspeaker L via the adaptive filter y(n), whereby the output signal y(n) of the adaptive filter $\hat{h}(n)$ is subtracted from the microphone signal d(n) at the summing block Σ_1 to generate the error signal e(n) for adapting the filter coefficients of the adaptive filter $\hat{h}(n)$.

[0032] By means of the adaptive filter

$$\hat{h}(n) = \left[\hat{h}_0(n), \hat{h}_1(n), \cdots, \hat{h}_{N-1}(n)\right]^T$$

an attempt is made to estimate the impulse response h(n) of the transfer path between a loudspeaker L and a microphone M, the feedback signal y(n) being estimated by convolving the loudspeaker signal x(n) with the estimated impulse response. The object of this estimation is to get a good matching of the estimated impulse response $\hat{h}(n)$ of the loudspeaker room microphone system with the real impulse response h(n) of the transfer path between loudspeaker L and microphone M. When this is the case, decoupling the system as a whole can be attained by deducting the estimated feedback (feedback signal y(n)) from the microphone signal d(n).

[0033] Referring now to FIG. 4 there is illustrated an assembly of an acoustic stereo echo compensator which uses two non-interdependent signals of a stereo signal source for echo compensation. FIG. 4 shows a stereo signal source S, a loudspeaker room microphone (LRM) system, two loudspeakers L1 and L2, a microphone M, two summing units Σ1 and Σ2 as well as two adaptive filters A1 and A2. The stereo signal source S generates two non-interdependent signals $X_1(z)$ and $X_2(z)$ serving as input signals for the loudspeakers L1 and L2 arranged in the loudspeaker room microphone (LRM) system. Likewise arranged in the loudspeaker room microphone (LRM) system is the microphone M. This results in an unknown acoustic transfer function $H_1(z)$ between the loudspeaker L1 and the microphone M as well as an unknown acoustic transfer function $H_2(z)$ between the loudspeaker L2 and the microphone M. Typically such transfer functions $H_1(z)$ and $H_2(z)$ depend on the actual configuration of a loudspeaker room microphone (LRM) system, such as, for example, the geometry, the existence and property of reflective surfaces and any furniture or seating arrangements (e.g. in a motor vehicle passenger compartment).

[0034] As evident from FIG. 4 the signal $X_1(z)$ serves as the input signal for the adaptive filter A1 and the signal $X_2(z)$ serves as the input signal for the adaptive filter A2. The adaptive filters A1 and A2 are used to approximate the transfer functions $H_1(z)$ and $H_2(z)$, for example, by using an NLMS algorithm. In this arrangement the output signals $Y_1(z)$ and $Y_2(z)$ of the adaptive filters A1 and A2 are added using the summing unit Σ1 and the negated result Y(z) is added to the output signal D(z) of the microphone M using the summing unit Σ2. This results in the error signal E(z) which is used in turn to optimize the filter parameters of the adaptive filters A1 and A2 in the sense of minimizing the error signal E(z). When, for instance, the microphone is arranged in a passenger compartment of a motor vehicle the microphone signal D(z) of the microphone M includes, in addition to the music signal components of the loudspeakers L1 and L2, also any background noise (for example, the noise of the vehicle on the move) existing and any speech signal components of the vehicle occupants.

[0035] Since the music signal components of the microphone signal D(z) are practically cancelled out by the assembly as shown in FIG. 4 only the signal components of the background noise and of any speech signals exist substantially in the signal E(z) which can thus be used, for example, to estimate the background noise and/or to further process speech signal components. Tests with an acoustic (two-channel) stereo echo compensator as shown in FIG. 4 have shown that, as compared, for example, to a single-channel echo compensator such as the one shown in FIG. 3, an increased reduction of the echo by roughly 10dB is achievable. The NLMS algorithm used equates as follows in the spectral domain:

$$Y(l,k) = \sum_{i=1}^{2} X_i(l,k) * \hat{H}_i(l,k) = X_1(l,k) * \hat{H}_1(l,k) + X_2(l,k) * \hat{H}_2(l,k) = Y_1(l,k) + Y_2(l,k)$$

where

Y(l, k) is the total (summed) output signal of the adaptive filters A1 and A2,
Y_1(l, k) and Y_2(l, k) are the corresponding output signals of the adaptive filters A1 and A2,
X_1(l, k) and X_2(l, k) are the corresponding input signals of the adaptive filters A1 and A2, and
$\hat{H}_1$(l, k) and $\hat{H}_2$(l, k) are the corresponding approximations of the transfer functions H_1(l, k) and H_2 (l, k).

[0036] The formula for computing the error signal is as follows:

$$e(n) = \begin{bmatrix} 0_{L\times 1} \\ d(n)_{L\times 1} \end{bmatrix} - W * F^{-1}\{Y(l, k)\}$$

where

W = diag{[0_{1xL}, 1_{1xL}]} is the rectangular window function, realizing for the so-called zero padding (to implement the cyclic convolution input sequence and impulse response must be of the same length). If not, the shorter of the two vectors must be correspondingly enlarged by adding zeros (also called zero padding),
F is the Fast Fourier Transformation (FFT),
F^{-1} is the Inverse Fast Fourier Transformation (IFFT),
$d(n)_{Lx1}$ is the microphone signal with length L (corresponding to half the FFT length respectively length of the adaptive filter(s),
0_{Lx1} is the zero vector with length L, and
e(n) is the error signal with length L in the time domain.

[0037] The formula for computing the filter coefficients by the NLMS algorithm is as follows:

$$\hat{H}(l, k+1) = \hat{H}(l, k) + \mu * S^{-1}(l, k) * X^{H}(l, k) * E_{b}(l, k)$$

resulting for the two filters A1 and A2

$$\hat{H}_1(l, k+1) = \hat{H}_1(l, k) + \mu * S^{-1}(l, k) * X_1^{H}(l, k) * E(l, k)$$

$$\hat{H}_2(l, k+1) = \hat{H}_2(l, k) + \mu * S^{-1}(l, k) * X_2^{H}(l, k) * E(l, k)$$

where

$$\hat{H}(l, k) = [\hat{H}_1(l, k)^T, \hat{H}_2(l, k)^T]^T,$$

μ is the adaptation step size, and

$$S(l,k) = \frac{1}{2} * \sum_{i=1}^{2} |X_i(l,k)|^2 = \frac{1}{2} * (|X_1(l,k)|^2 + |X_2(l,k)|^2) =$$
$$\frac{1}{2} * (\mathrm{Re}(X_1(l,k))^2 + \mathrm{Im}(X_1(l,k))^2 + \mathrm{Re}(X_2(l,k))^2 + \mathrm{Im}(X_2(l,k))^2)$$

whereby S(l, k) is the vector containing the mean value of all spectral energy values of the input signals,

X(l, k) = [X_1 (1, k), X_2 (1, k)] is the input signal matrix
E_b(l, k) = [E(l, k)^T, E(l, k)^T] is the spectral block error signal, and
E(l, k) = F{e(n) } is the spectral error signal.

with T = transponse; H = conjugate, complex transpose (= Hermitian transpose).

[0038] It is currently usual to employ assemblies in motor vehicle sound systems featuring a plurality of loudspeakers for replicating a corresponding plurality of music signals as surround sound systems. In such assemblies providing music signals via correspondingly assigned non-interdependent loudspeakers a corresponding number of different transfer functions is formed between the corresponding loudspeakers and a microphone arranged in the LRM system. Thus, an assembly for suppressing music signal components in a microphone signal may also comprise a plurality of adaptive filters for approximating all or some of the plurality of transfer functions.

[0039] Referring now to FIG. 5 there is illustrated in a block circuit diagram a prior art assembly of a multichannel active matrix decoding system for stereo input signals, known as a logic7® decoding system. FIG. 5 shows a logic7® matrix decoder 1, eight signal amplifier units 2, 3, 4, 5, 6, 7, 8 and 9 and eight loudspeakers 10, 11, 12, 13, 14, 15, 16 and 17. The logic7® matrix decoder 1 comprises two signal inputs 18 and 19 for stereo input signals 20 and 21, the signal input 18 serving to receive the stereo input signal 20 of the left channel of a dual channel stereo signal and signal input 19 serving to receive the stereo input signal 21 of the right channel of a dual channel stereo signal. The logic7® matrix decoder 1 comprises furthermore eight signal outputs for the signals 22, 23, 24, 25, 26, 27, 28 and 29.

[0040] As evident from FIG. 5 the logic7® matrix decoder 1 generates from stereo input signals 20 (left stereo channel) and 21 (right stereo channel) non-interdependent signals 22, 23, 24, 25, 26, 27, 28 and 29 which are amplified by corresponding output signal amplifier units 2, 3, 4, 5, 6, 7, 8 and 9 and forwarded to corresponding loudspeakers 10, 11, 12, 13, 14, 15, 16 and 17 of a multichannel audio system. In this arrangement the amplified signal 22 serves to activate the loudspeaker 10 which in a given room corresponds to a left front loudspeaker situated front, left relative to the position of a listener. The amplified signal 24 serves to activate the loudspeaker 12 which in a given room corresponds to a right front loudspeaker situated front, right relative to the position of a listener. The amplified signal 23 serves to activate the loudspeaker 11 which in a given room corresponds to a center loudspeaker situated between the left front loudspeaker 10 and the right front loudspeaker 12.

[0041] The amplified signals 25, 26, 27, 28 and 29 correspondingly serve to activate the loudspeakers 13, 14, 15, 16 and 17. In this arrangement the loudspeaker 13 is positioned left relative to a listener and loudspeaker 14 positioned right relative to a listener whilst loudspeaker 15 is positioned rear left relative to a listener and loudspeaker 16 is positioned rear right relative to a listener. The signal 29 amplified by the signal amplifier unit 9 serves to activate the subwoofer 17 which exclusively serves to replicate the low-frequency signal components of the audio signal, it making no contribution to the surround effect of replication created by the loudspeakers 10, 11, 12, 13, 14, 15 and 16.

[0042] Referring now to FIG. 6 there is illustrated a block circuit diagram of an assembly of a seven-channel audio system as shown in FIG. 5 in a given room which may be, for example, the passenger compartment of a motor vehicle. Relative to listeners 30 and 31, FIG. 6 shows a left front loudspeaker 10, a right front loudspeaker 12, a loudspeaker 11 positioned in the middle between the left and right front loudspeakers, a side loudspeaker 13 positioned left, a right side loudspeaker 14, a left rear loudspeaker 15 and a right rear loudspeaker 16. Not shown in the example assembly of FIG. 6 is the subwoofer 17 likewise usually included in a seven-channel audio system.

[0043] Evident furthermore from FIG. 6 are six signal processing blocks 32, 33, 34, 35, 36 and 37 as components of the logic7® matrix decoder 1 as shown in FIG. 5 to generate the corresponding signals 22, 23, 24, 25, 26, 27, 28 and 29 for activating the loudspeakers 10, 11, 12, 13, 14, 15, 16 and 17 (see FIG. 5). Signal components for the left front loudspeaker 10 and for the right front loudspeaker 12 are used in the logic7® matrix decoder 1 to generate therefrom the signal for the center loudspeaker 11 (as detailed below). In this arrangement the signal processing blocks 32 and 33 serve to attenuate the amplitude of these signal components as a function of their spectral distribution and as a function of the wanted surround effect. The attenuation in a logic7® matrix decoder 1 is usually in the range 0 dB to -7.5 dB.

[0044] The signal processing blocks 34, 35, 36 and 37 serve to delay in time the signals generated from both stereo input signals (see signals 20 and 21 as shown in FIG. 5) for the loudspeakers 13, 14, 15 and 16 for the wanted surround reverberation effect) and to shelve the level in certain frequency bands (surround effect) as is usually done by using roll-off and shelving filters.

[0045] Shelving the frequency bands of the original stereo input signal and delaying in time the effect defines the surround sound effect and the reverberation time. Rolling off the high-frequency components in the signals replicated by the loudspeakers 13, 14, 15 and 16 shifts the sound upfront, for instance. Such a surround system features an adjustable time delay between the sound signals replicated by the left front loudspeaker 10 and by the side left loudspeaker 13 also termed surround loudspeaker. This time delay is effected by the signal processing block 34 and amounts to roughly 8 ms as is usual for motor vehicle sound system applications, this likewise applying to the time delay between the right front loudspeaker 12 and the side right surround loudspeaker 14 as effected by the signal processing block 35.

[0046] In addition to this, such a surround system features a further adjustable time delay between the sound signals replicated by the side left loudspeaker 13 and by the rear left loudspeaker 15. This time delay is effected by the signal processing block 36 and amounts to roughly 14 ms as is usual for motor vehicle sound system applications, this likewise applying to the time delay between the side right loudspeaker 14 and the rear right loudspeaker 16 as effected by the signal processing block 37. Not shown in FIG. 6 is an optional subwoofer as may likewise be included.

[0047] The object of a matrix decoder such as for example the logic7® matrix decoder 1 as shown in FIG. 5 is to convert signals from, for example, two input channels (stereo signals) into, for example, 7 output channels to create the wanted stereo surround effect in a given room. These output channels are used to activate loudspeakers located in various positions in the room (see FIG. 6) By being processed accordingly in an active matrix decoder such as the logic7 matrix decoder the signals intended to come acoustically from a certain direction are processed in the matrix decoder so that they appear to come from the corresponding direction for the listener when replicated by the loudspeakers of the audio system, thus defining for a certain point in time what is called a listener event direction and, where necessary, the location of such a listener event, both of which can change with time in a dynamic audio signal.

[0048] In this arrangement the output signals of a matrix decoder are linear combinations of two input signals (stereo signal). In an active matrix decoder such as the logic7® matrix decoder the coefficients of the linear combinations (the matrix elements) are functions of time which change non-linearly but slowly as compared to the audible frequencies. These matrix elements may also be complex functions of the frequency and time. It is thus the task of such a decoder to define or control the response of these coefficients.

[0049] The simplest matrix decoder is a passive matrix decoder in which all coefficients are fixed values wherein the output signal for a left loudspeaker results from the input signal for the left channel multiplied by 1, the output signal for a center loudspeaker resulting from the input signal for the left channel multiplied by 0.7 plus the input signal for the right channel multiplied by 0.7 and the output signal for a right loudspeaker resulting from the input signal for the right channel multiplied by 1.

[0050] By contrast, the demands on an active matrix decoder, such as for example the logic7® matrix decoder, are significantly more far-reaching, also influencing the signal generated for the center loudspeaker. This is particularly the case when a strongly directed signal (for instance a signal component intended to be output substantially in the left portion of the replication room by a surround system) is a component of the stereo input signal.

[0051] When no uncorrelated (non-directed) signal component exists in the input signals, the channels replicating no directed signal component are required to comprise only a minimum output signal. Thus, for example, a signal required to appear in the middle between a right loudspeaker and a center loudspeaker in stereo output is required not to generate any output signals for the left and rear loudspeaker of a multichannel audio system. In the same way, a signal intended to be output in the middle is required not to comprise signal components for left and right loudspeakers. Furthermore, the total output signal of the decoder is required to create the same impression for the volume when the motion of a directed signal is in various portions of the surround.

[0052] Also, with any change in the matrix elements of the decoder the total energy of the non-directed signal component of an audio signal must be maintained constant in every output channel to replicate a directed signal changing in direction. In addition to this the transition between replication of signal components all non-directed and signal components all directed must be even with no shifts in the perceived direction of the sound rendering. All of these requirements are satisfied by the logic7® matrix decoder and the signals for the corresponding loudspeakers, such as the center loudspeaker of a surround system are conditioned accordingly.

[0053] Further examples of assemblies handling multichannel non-interdependent music signals are, for example, replication systems providing discrete signals for activating loudspeakers in accordance with "Dolby Digital 5.1® " or "DTS 6.1 discrete" operating 6 and 7 loudspeakers respectively in an LRM system of non-interdependent activation signals.

[0054] Where multichannel sound systems based for example on "Logic7® ", "Dolby Digital 5.1® " or "DTS 6.1 discrete" are concerned, the assembly as shown in FIG. 4 can now be expanded so that also the transfer functions configured between more than two loudspeakers and a microphone arranged in the LRM system are approximated by corresponding adaptive filters.

[0055] Referring now to FIG. 7 there is illustrated an assembly of an acoustic echo compensator (AEC) using, for example, four non-interdependent signals of a signal source for echo compensation which could also use more than four signals, however, since the signal source comprises seven non-interdependent output signals as is the case, for instance with a logic7® decoder. Evident from FIG. 7 are a signal source S, a loudspeaker room microphone (LRM) system, seven loudspeakers L1, L2, L3, L4, L5, L6 and L7 arranged in the loudspeaker room microphone(LRM) system, a microphone 7, two summing units Σ1 and Σ2 as well as four adaptive filters A1, A2, A3 and A4.

[0056] The signal source S generates seven non-interdependent signals $X_1(z)$, $X_2(z)$, $X_3(z)$, $X_4(z)$, $X_5(z)$, $X_6(z)$ and $X_7(z)$ serving as input signals for the loudspeakers L1, L2, L3, L4, L5, L6, and L7 arranged in the loudspeaker room microphone (LRM) system, also featuring a microphone M. This results in an unknown acoustic transfer function $H_1(z)$, $H_2(z)$, $H_3(z)$, $H_4(z)$, $H_5(z)$, $H_6(z)$ and $H_7(z)$ being configured respectively between the loudspeaker L1, L2, L3, L4, L5, L6, L7 and the microphone M. As explained with reference to FIG. 4 these transfer functions $H_1(z)$, $H_2(z)$, $H_3(z)$, $H_4(z)$, $H_5(z)$, $H_6(z)$ and $H_7(z)$ depend, in turn, on the actual configuration of the LRM system, such as, for example, its geometry, any reflecting surfaces and their response and any furniture or seating arrangements (in a motor vehicle passenger compartment, for instance).

[0057] Referring still to FIG. 7 there is illustrated how the signal $X_1(z)$, $X_2(z)$, $X_3(z)$ and $X_4(z)$ respectively serves as the input signal for the adaptive filter A1, A2, A3 and A4. By the adaptive filters A1, A2, A3 and A4 the transfer functions H1(z), $H_2(z)$, $H_3(z)$ and $H_4(z)$ are approximated, for example in turn by using an NLMS algorithm. In this arrangement the output signals $Y_1(z)$, $Y_2(z)$, $Y_3(z)$ and $Y_4(z)$ of the adaptive filters A1, A2, A3 and A4 are added using the summing unit Σ1 and the negated result Y(z) is added to the output signal D(z) of the microphone M using the summing unit Σ2.

[0058] This results in the error signal E(z) which is used in turn to optimize the filter parameters of the adaptive filters A1, A2, A3 and A4 in the sense of minimizing the error signal E(z). When, for instance, the microphone M is arranged in the passenger compartment of a motor vehicle the microphone signal D(z) of the microphone M picks up, in addition to the music signal components of the loudspeakers L1, L2, L3, L4, L5, L6 and L7, also the existing background noise (for example the noise of the vehicle on the move) and any speech signal components of the vehicle occupants. An assembly is shown in FIG. 7 in which four transfer functions are approximated and the corresponding signal components in the microphone signal D(z) cancelled, although, of course, further existing transfer functions may be included in the processing.

[0059] The NLMS algorithm used can be equated as follows for a multichannel assembly in the spectral range:

$$Y(l,k) = \sum_{i=1}^{n} X_i(l,k) * \hat{H}_i(l,k) = X_1(l,k) * \hat{H}_1(l,k) + \ldots + X_n(l,k) * \hat{H}_n(l,k) = Y_1(l,k) + \ldots + Y_n(l,k)$$

where

Y(l, k) is the total (summed) output signal of the adaptive filters A1 ... An,
Y_1(l, k) ... Y_n(l, k) are the corresponding output signals of the adaptive filters A1 ... An,
X_1(l, k) ... X_n(l, k) are the corresponding input signals of the adaptive filters A1... An,
$\hat{H}_1$(l, k) ... $\hat{H}_n$(l, k) are the corresponding approximations of the transfer functions H_1(1, k) ... H_n(l, k).

[0060] The formula for computing the error signal is the same as that described in conjunction with FIG 4.

[0061] Computing the filter coefficients by the NLMS algorithm is done in accordance with the formula

$$\hat{H}(l,k+1) = \hat{H}(l,k) + \mu * S^{-1}(l,k) * X^H(l,k) * \tilde{E}_b(l,k)$$

resulting in, for the filters A1 ... An:

$$\hat{H}_1(l,k+1) = \hat{H}_1(l,k) + \mu * S^{-1}(l,k) * X_1^H(l,k) * \tilde{E}(l,k)$$

$$\ldots\ldots.$$

$$\hat{H}_n(l,k+1) = \hat{H}_n(l,k) + \mu * S^{-1}(l,k) * X_n^H(l,k) * \tilde{E}(l,k)$$

where n = 4 for the assembly as exemplarily shown in FIG. 7.

[0062] Each loudspeaker may be substituted by a group of loudspeakers such as, e.g., a tweeter, a midrange speaker and a woofer connected together e.g. by a passive or active filter network.

[0063] As shown above, the assembly acoustically replicates the non-interdependent activation signals of the multichannel music signals as music signals via the dedicated loudspeakers arranged in the LRM system, and converts via the microphone arranged in the LRM system the total sound level existing in situ at the microphone into a corresponding microphone signal. Furthermore, the assembly approximates the acoustic transfer functions configured in the LRM system between the loudspeakers and the microphone for each of the activation signals by each of the dedicated adaptive filter units, and filters the activation signals of the multichannel music signal source assigned to the corresponding acoustic transfer functions with the corresponding adaptive filter units approximating each of the acoustic transfer functions.

[0064] Although various examples to realize the invention have been disclosed, it will be apparent to those skilled in the art that various changes and modifications can be made which will achieve some of the advantages of the invention without departing from the spirit and scope of the invention. It will be obvious to those reasonably skilled in the art that other components performing the same functions may be suitably substituted. Such modifications to the inventive concept are intended to be covered by the appended claims.

Claims

1. A loudspeaker-room-microphone (LRM) system comprising:

   a room in which $L \geq 2$ loudspeakers and a microphone that provides an output signal are arranged;
   a multichannel signal source configured to generate L non-interdependent activation signals each supplied to one of the L groups of loudspeakers;
   M adaptive filters each of which is supplied with one of the L activation signals where $L \geq M \geq 2$; the adaptive filters have filter transfer functions, receive an error signals and supply output signals;
   a calculation unit that is connected downstream of the M adaptive filters and the microphone and that sums up the output signals of the adaptive filters and subtracts the sum from the output signal of the microphone or vice versa; the calculation unit supplies to the M adaptive filters the error signal that also serves as an output signal of the LRM system; and
   LRM transfer functions representing signal paths between each of the loudspeakers and the microphone; where each adaptive filter has a filter transfer function that approximates one of the LRM transfer function.

2. The system as set forth in claim 1 where
   the multichannel signal source provides more than two non-interdependent activation signals for a corresponding plurality of dedicated loudspeakers arranged in the LRM system, resulting in a plurality of acoustic transfer functions being configured, and where
   more than two acoustic transfer functions configured in the LRM system between loudspeakers and the microphone are each approximated by a dedicated adaptive filter unit.

3. The system as set forth in claim 1 or 2 where the non-interdependent activation signals include a stereo music channel signal.

4. The system as set forth in claim 1 or 2 where the multichannel music signal source provides activation signals configured by surround sound algorithms from a stereo signal.

5. The system as set forth in any of the claims 1 or 2 wherein the multichannel music signal source is a logic7 music signal source.

6. The system as set forth in claim 1 or 2 wherein the multichannel music signal source is a Dolby Digital or DTS music signal source.

7. The system as set forth in any of the claims 1 or 2 wherein the LRM system is the passenger compartment of a motor vehicle.

8. A method for reducing music signal components in an output signal of at least one microphone arranged in a loudspeaker-room-microphone (LRM) system in which $L \geq 2$ loudspeakers; the method comprises the steps of:

   generating by means of a multichannel signal source at least two non-interdependent activation signals for the loudspeakers arranged in the LRM system;
   acoustically replicating the non-interdependent activation signals via the loudspeakers;
   converting the total sound level existing in situ at the microphone of the LRM system into a corresponding microphone signal by the microphone arranged in the LRM system;
   approximating the acoustic transfer functions configured in the LRM system between the loudspeakers and the microphone for each of the activation signals by each assigned adaptive filter unit;
   filtering the activation signals assigned to the corresponding acoustic transfer functions by the corresponding adaptive filter units approximating each acoustic transfer function; and
   adding the output signals of the adaptive filter units to a sum signal, and subtracting thereof the microphone signal.

9. The method as set forth in claim 8 further comprising the steps of:

providing more than two non-interdependent activation signals, resulting in a plurality of acoustic transfer functions being configured; and
approximating more than two acoustic transfer functions in the LRM system between the loudspeakers and the microphone each by a dedicated adaptive filter unit.

10. The method as set forth in claim 8 or 9 where the non-interdependent activation signals include a stereo music channel signal.

11. The method as set forth in claim 10 where more than two acoustic transfer functions configured in the LRM system between loudspeakers and the microphone are each approximated by the adaptive filter unit.

12. The method as set forth in claim 10 or 11 where the multichannel music signal source provides activation signals configured by surround sound algorithms from a stereo signal.

13. The method as set forth in any of the claims 10 to 12 where the multichannel signal is provided by a Logic7 music signal source.

14. The method as set forth in claim 10 or 11 where the multichannel signal source is provided by a Dolby Digital or DTS music signal source.

15. The method as set forth in any of the claims 8 to 14 where the LRM system is the passenger compartment of a motor vehicle.

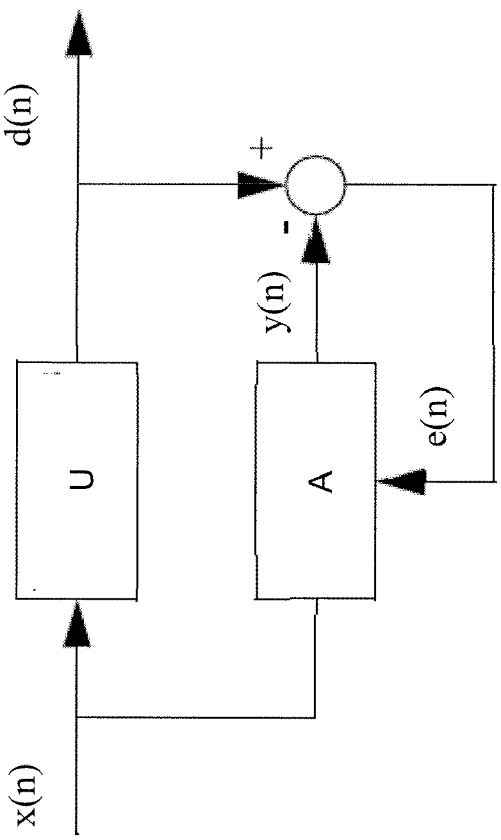

FIG. 1

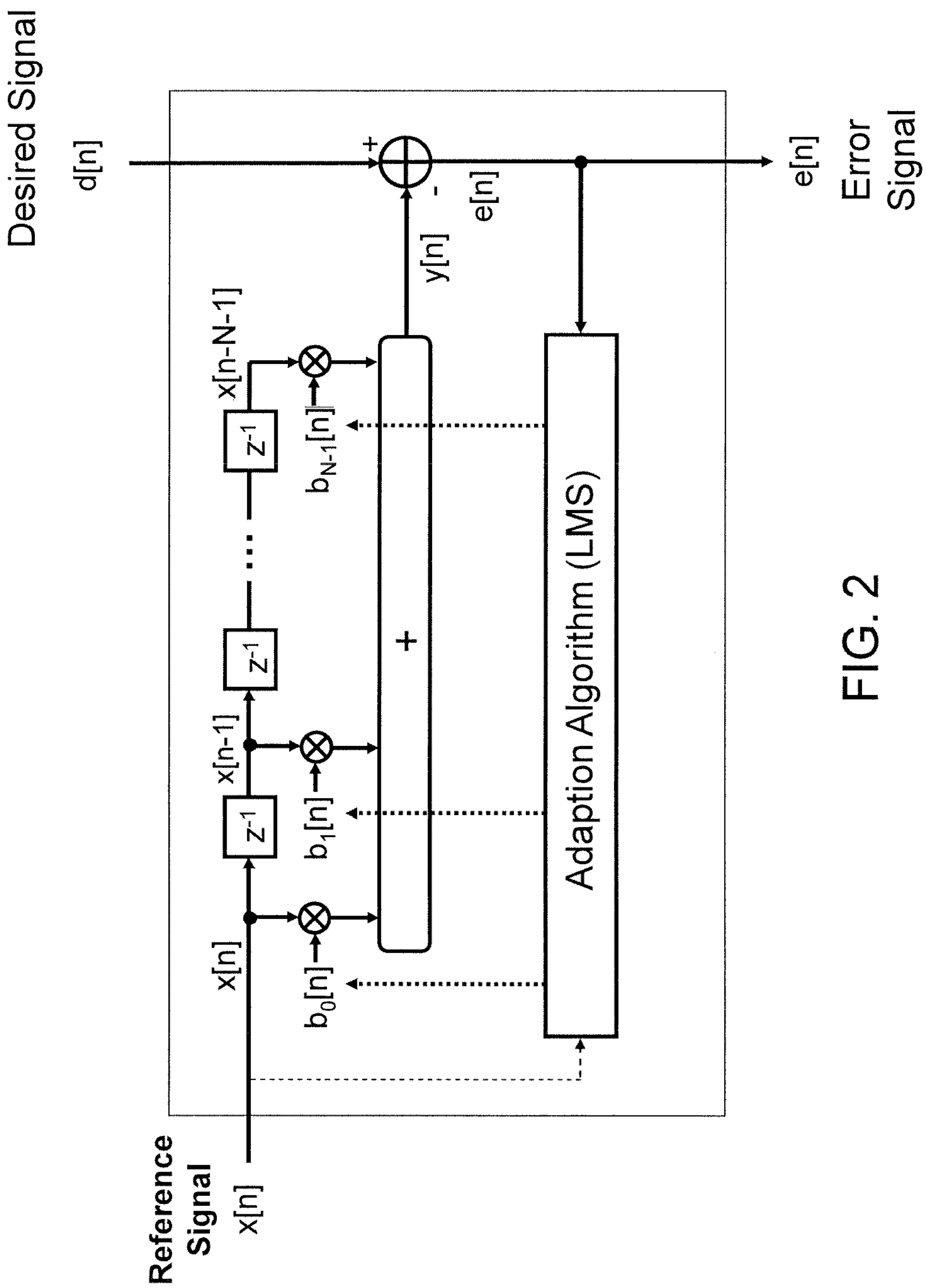
Desired Signal
d[n]
+
-
e[n]
y[n]
x[n-N-1]
z-1
z-1
z-1
bN-1[n]
+
Adaption Algorithm (LMS)
x[n-1]
b1[n]
x[n]
b0[n]
Reference Signal
x[n]
e[n]
Error Signal

FIG. 2

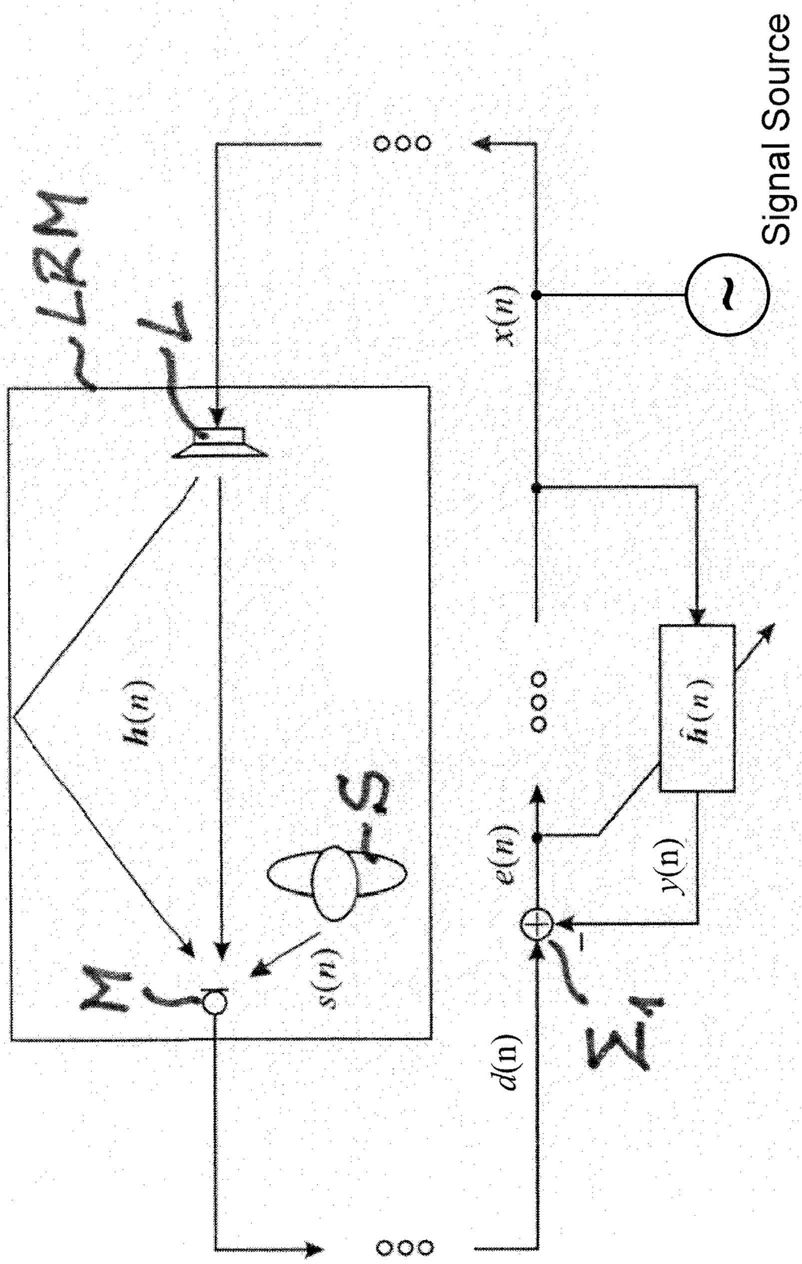

FIG. 3

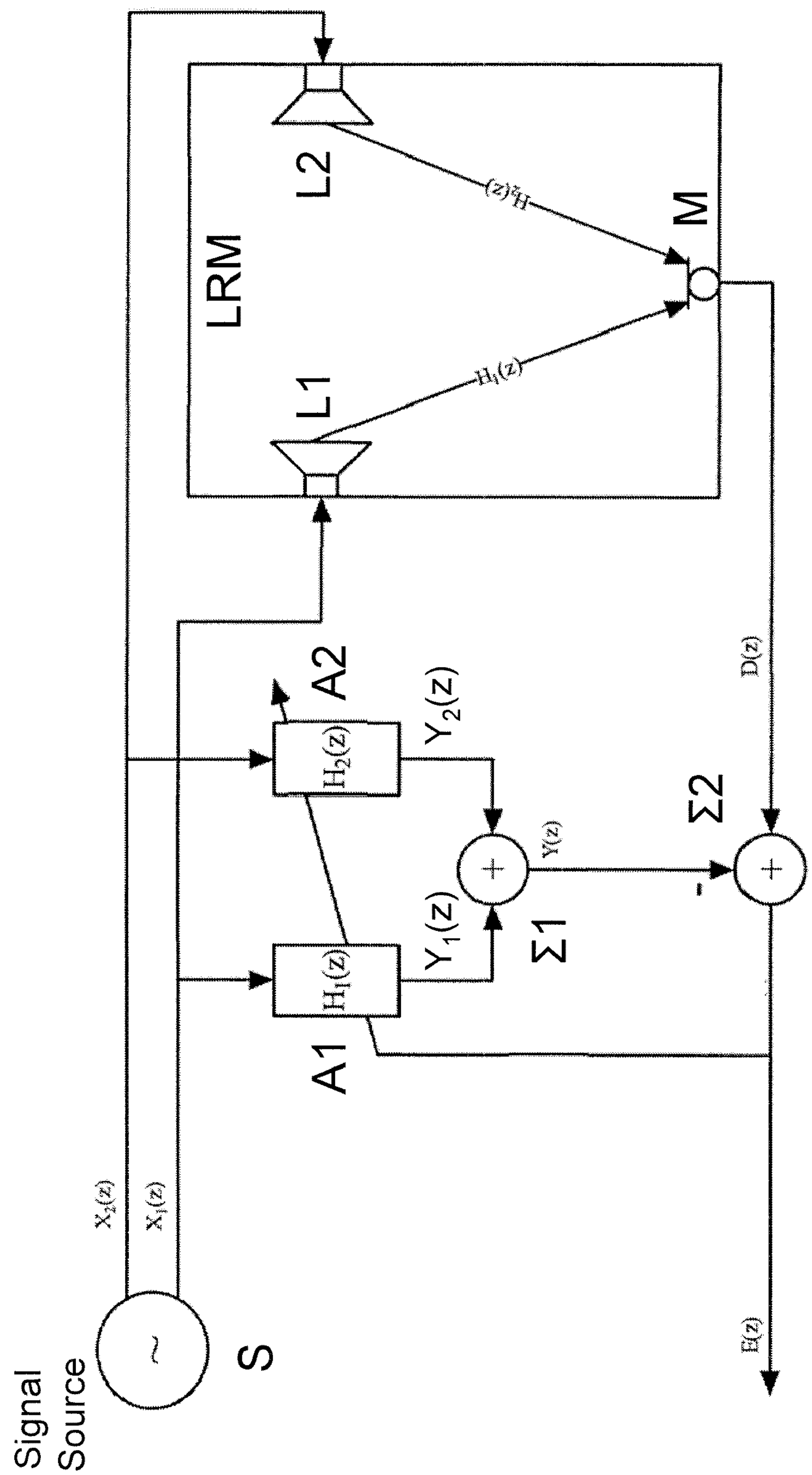
LRM
L1
L2
M
$H_1(z)$
$H_2(z)$
Signal Source
S
$X_2(z)$
$X_1(z)$
A1
A2
$H_1(z)$
$H_2(z)$
$Y_1(z)$
$Y_2(z)$
Σ1
Σ2
Y(z)
D(z)
-
E(z)

FIG. 4

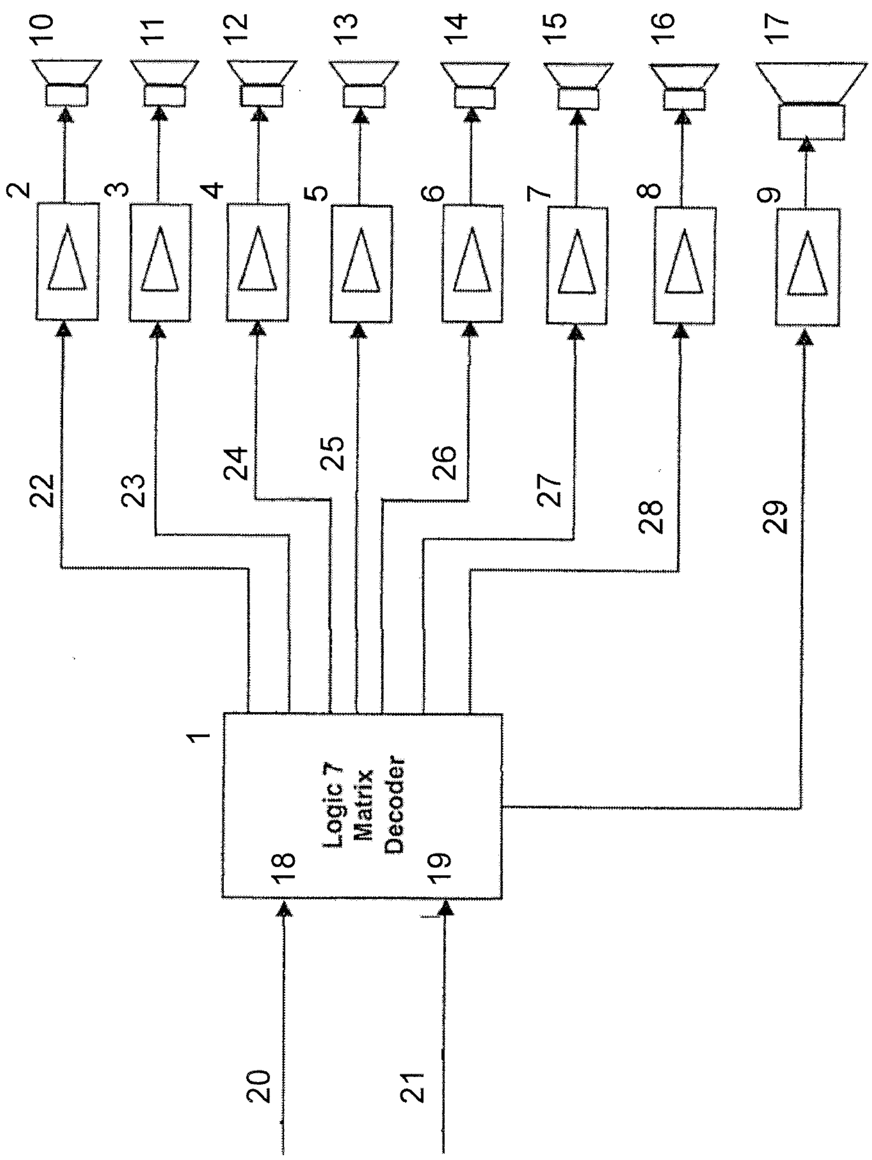
10
11
12
13
14
15
16
17
2
3
4
5
6
7
8
9
22
23
24
25
26
27
28
29
1
Logic 7
Matrix
Decoder
18
19
20
21

FIG. 5

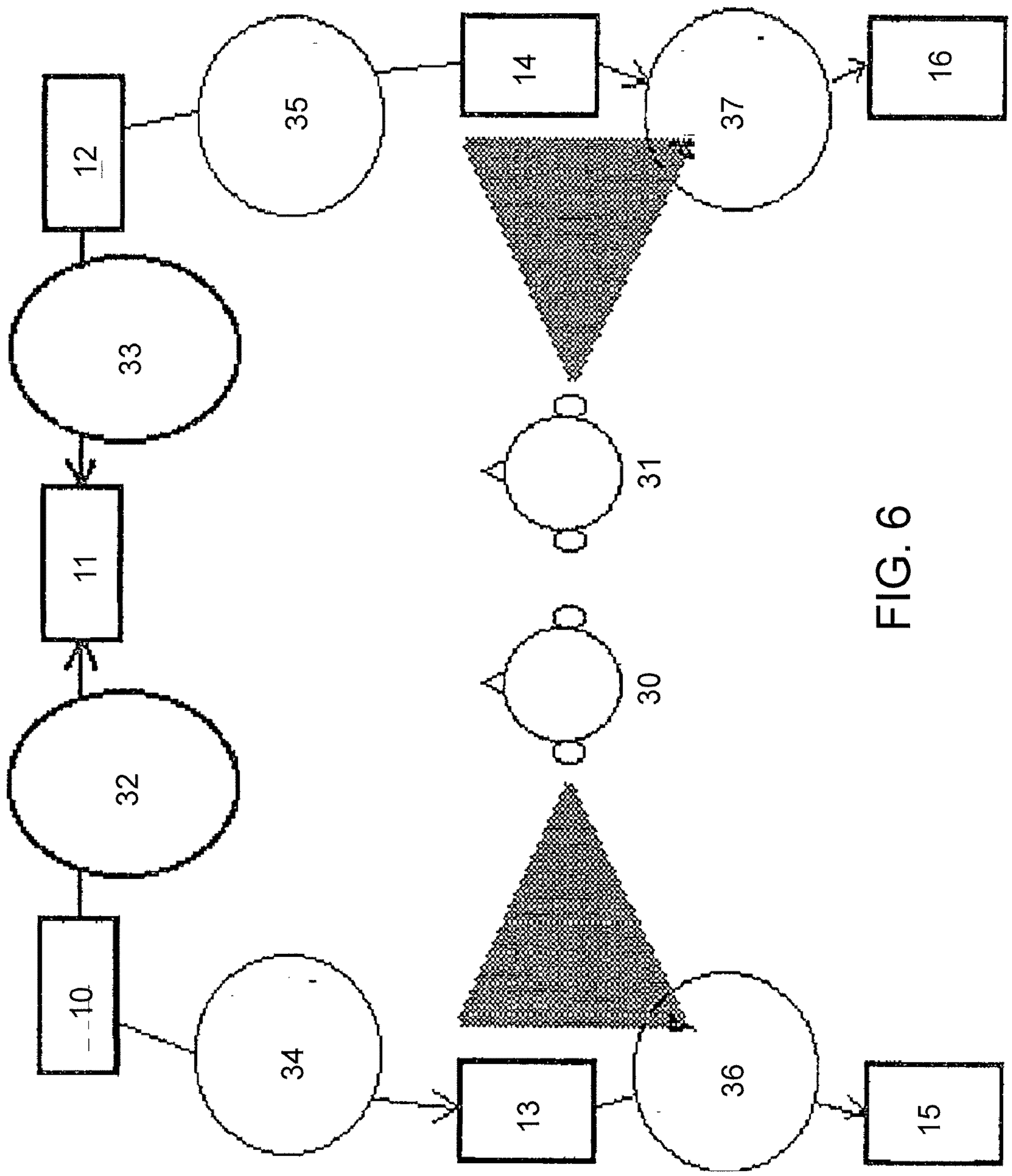

FIG. 6

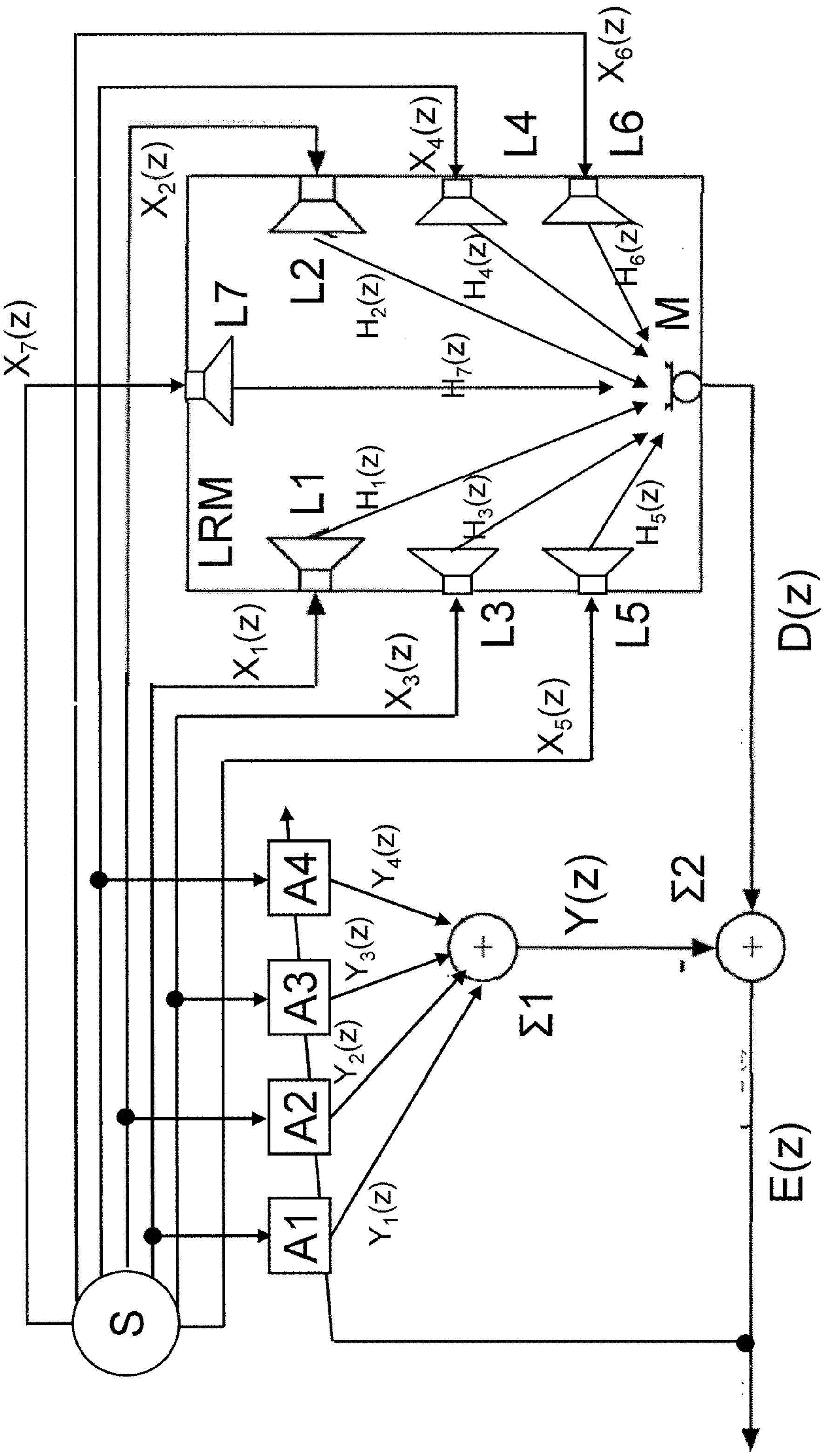
S
A1
A2
A3
A4
Y1(z)
Y2(z)
Y3(z)
Y4(z)
Σ1
Y(z)
Σ2
E(z)
D(z)
LRM
L1
L2
L3
L4
L5
L6
L7
M
X1(z)
X2(z)
X3(z)
X4(z)
X5(z)
X6(z)
X7(z)
H1(z)
H2(z)
H3(z)
H4(z)
H5(z)
H6(z)
H7(z)

FIG. 7

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EUROPEAN SEARCH REPORT

Application Number
EP 09 16 1443

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (IPC)
X	WO 03/061344 A2 (KONINKL PHILIPS ELECTRONICS NV [NL]; SCHOBBEN DANIEL W E [NL]; EGELMEE) 24 July 2003 (2003-07-24) * page 1, line 5 * * page 3, line 23 - line 31 * * page 4, line 25 - line 34 * -----	1-15	INV. H04R27/00 H04S3/02 H04M9/08
A	EP 1 475 781 A2 (ALPINE ELECTRONICS INC [JP]) 10 November 2004 (2004-11-10) * paragraph [0012] * -----	1,8	

TECHNICAL FIELDS SEARCHED (IPC)

H04R
H04S
H04M

The present search report has been drawn up for all claims

4

Place of search	Date of completion of the search	Examiner
Munich	8 December 2009	Heiner, Christoph

CATEGORY OF CITED DOCUMENTS

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08-12-2009

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
WO 03061344 A2	24-07-2003	AU 2002367065 A1	30-07-2003
		CN 1636423 A	06-07-2005
		JP 2005515737 T	26-05-2005
		US 2005070243 A1	31-03-2005
EP 1475781 A2	10-11-2004	CN 1542734 A	03-11-2004
		JP 4209247 B2	14-01-2009
		JP 2004333704 A	25-11-2004
		US 2004260549 A1	23-12-2004

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