(54) **System and method for improving communication in a room**

(57)   System and method for improving the acoustical communication between interlocutors in at least two positions in a room, comprising generating electrical signals representative of acoustical signals present at the respective interlocutor positions; amplifying each of said electrical signals; and converting said amplified electrical signals into acoustical signals; wherein said electrical signals are each delayed with a delay time such that the acoustical signal arriving first at one of the interlocutor positions originates from the direction of the other interlocutor position.
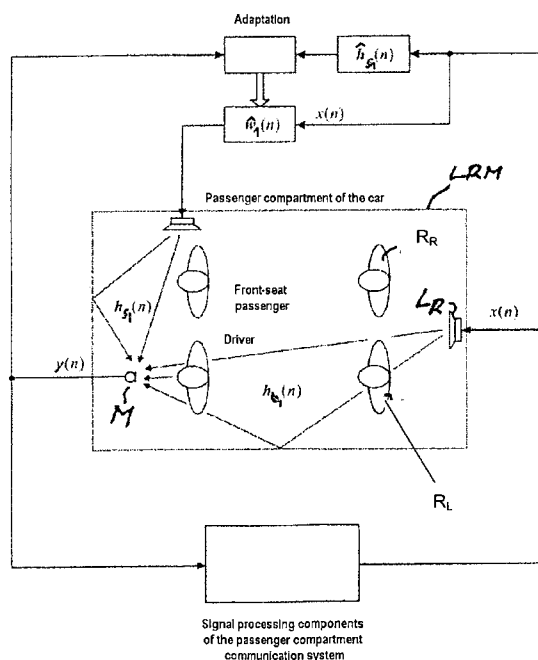
Fig. 5

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Description

TECHNICAL FIELD

[0001]   The invention relates to a system and method for improving communication in a room and in particular to a method for suppressing feedback and improving the perception of direction in room communication systems, for example, passenger compartment communication systems of motor vehicles.

BACKGROUND

[0002]   In order to improve speech comprehensibility in motor vehicles, so-called passenger compartment communi-cation systems are used. Such systems are capable of improving the speech quality and the speech comprehensibility when conversations are being conducted in the moving motor vehicle, that is, e.g., in the case of the simultaneous effect of motion noise from the motor vehicle itself or external noise sources in the vehicle's surroundings. This applies, in particular, when one of the participants (interlocutors) in the conversation is in one of the front seats and another participant is in one of the rear seats and there is an overall high level of noise. Fig. 1 illustrates an overview of such a system.

[0003]   In this case, the block diagram of the arrangement of a passenger compartment communication system as shown in Fig. 1 comprises a loudspeaker-room-microphone system LRM which, as in the present case, may be the passenger compartment of a car. The loudspeaker-room-microphone system LRM has, by way of example, four seating positions for passengers, which are designated driver, front-seat passenger, rear left seating position R_L and rear right seating position R_R. Depending on the design of the car, additional seats or additional rows of seats may also be present. The loudspeaker-room-microphone system LRM shown in Fig. 1 also comprises loudspeakers L_{FL} (front left), L_{FR} (front right), L_{RL} (rear left) and L_{RR} (rear right) which form the sound reproduction system of the exemplary passenger com-partment communication system.

[0004]   In real applications, passenger compartment communication systems, particularly in luxury cars, may be of very much more complex design and typically comprise a multiplicity of loudspeakers and groups of loudspeakers at a wide variety of positions in the passenger compartment, use also typically being made, inter alia, of loudspeakers and groups of loudspeakers for different frequency ranges (for example subwoofers, woofers, medium-tone speakers and tweeters etc.). As shown in Fig. 1, the exemplary loudspeaker-room-microphone system LRM also comprises a multiplicity of microphones which are respectively assigned in groups to the seating positions for the passengers; by way of example, there are two respective microphones for each seat in Fig. 1. Using a plurality of microphones for each seating position allows, for example, to optimize the directivity of recorded speech signals for the respective seating position and thus to optimize the sound source which is to be recorded.

[0005]   Appropriate signal processing components (illustrated in Fig. 1 to the left of the loudspeaker-room-microphone system LRM) may be used to filter, amplify, attenuate, or change the phase angle of or temporally delay, inter alia, the speech signals recorded at the different seating positions using the microphones or groups of microphones, before they are reproduced using the passenger compartment communication system, in order to achieve the respective desired auditory impression. In particular, the speech signals traveling from the rear to the front and from the front to the rear are treated differently in this case (see signal processing components in Fig. 1).

[0006]   Using such systems for passenger compartment communication, the speech signal of the person who is speak-ing at the time is recorded using one or more microphones assigned to this person's seat and, after appropriate signal processing, is reproduced using those on-board loudspeakers of the passenger compartment communication system which are situated in the vicinity of the remaining passengers (in this case, a typical passenger compartment commu-nication system comprises a multiplicity of loudspeakers or groups of loudspeakers which are respectively arranged, for example, on the front, middle and rear sides and, if appropriate, also additionally in the centre of the passenger compartment of a motor vehicle and can be individually controlled). A fundamental disadvantage of such a method is that the acoustic localization and the visual localization of the speaker do not match in this case, particularly for passengers who are in rows of seats other than that of the respective speaker (for example, the speaker in the driver's seat, and the listener in one of the rear seats), since the speech signal of the speaker is predominantly received from loudspeakers which are respectively situated in the immediate vicinity of the listener. In addition, without appropriate signal processing of these speech signals, which is interposed between the recording and reproduction of the speech signals, such a system may become unstable on account of acoustic feedback as undesirable feedback noise, for example whistling, which may be very loud, no longer decays and is reproduced using the loudspeakers of the passenger compartment communication system may occur.

[0007]   If a plurality of microphones are assigned to each seat in the corresponding passenger compartment commu-nication system for the purpose of recording the speech signals, a beamformer output signal is first of all calculated from this plurality of microphone signals for each of these seats. Before being reproduced using the loudspeakers of the

produced. In this case, a delay of approximately 2 ms is sufficient to avoid this undesirable behaviour when filtering speech signals. In addition, on account of the periodicity of speech signals, the "memory" of an adaptive FIR filter ($\alpha = 0$ in Fig. 2) must not be too large, in particular it must not be selected to be larger than the reciprocal of the speech fundamental frequency to be expected. For this reason, the filter should comprise no more than 80 to 120 coefficients or samples N_c (at a sampling rate of 16 kHz) which are used for the calculation.

[0016] Since speech signals also contain components which have been correlated in short time ranges, the adaptive filter structure shown in Fig. 2 first of all also tries to suppress these components. This undesirable behaviour may be largely prevented if only a small maximum permissible step size μ is permitted for the change in the filter coefficients during adaptation. In this case, only those periodic signal components which are present in the speech signal for a relatively long period of time are removed. On the other hand, however, a small step size also results in slow convergence, that is to say slow adaptation of the adaptive filter to rapid changes in the signal to be processed. Therefore, sudden interference is also suppressed only after a period of time which cannot be ignored and can be perceived by human hearing. For this reason, an appropriate compromise must be included in the step size μ for changing the filter coefficients during adaptation in order to obtain an acoustic signal which is optimized with respect to human hearing sensitivities for a range of realistic ambient conditions which is as wide as possible. In this case, step sizes μ in the range of from 0.00001 to 0.01 have proved to be expedient for the exemplary case of using the NLMS algorithm for adaptively adapting the FIR filter.

[0017] The FIR structure of the feedback suppression filter may be extended using a weighted feedback path (see Fig. 2). Varying the feedback gain α makes it possible, in the extreme case, to convert the filter from a pure FIR structure ($\alpha = 0$) to a pure oscillator ($\alpha = 1$), it also being possible to select any desired values α between 0 and 1 (IIR filter). Inserting the feedback path is motivated by the fact that an attempt is made to profit from the advantages of a noise compensator having a periodic reference signal. The extension makes it possible to implement considerably more narrowband attenuation than with a pure FIR structure. On the other hand, the adaptive behaviour of the filter may result in an unstable filter being produced (see IIR filter). In order to prevent this, complicated stability tests must be carried out in such a case after each adaptation step. When implemented in real applications, only the FIR filter structure ($\alpha = 0$) is therefore frequently used in order to avoid instability in the filter structure.

[0018] In addition, adaptive feedback suppression filters have another quite considerable disadvantage. As soon as oscillation is detected at a particular frequency, the adaptive filter will attenuate the signal components at this frequency as determined. As a result, the levels of the spectral components which are responsible for the feedback are reduced in the loudspeaker signal u(n) to such an extent that feedback no longer occurs, which, for the time being, represents the desired behaviour. This suppression consequently also results in the feedback initially disappearing from the microphone signal, as desired. However, this in turn results in the attenuation of the signal components being adaptively reversed again in the relevant frequency range and in the feedback gaining power again. As soon as this has happened, the adaptive filter adjustment process begins again for these spectral components, and a type of oscillation of the attenuation response of the adaptive filter consequently results. Although feedback is suppressed in this manner, this does not take place durably or continuously to the desired extent.

[0019] Conventionally, use is therefore made of a further arrangement and a further method for reducing feedback. These are so-called compensation filters which have similar functional features to echo compensation in hands-free telephones. The structure of such an arrangement is illustrated, by way of example, in Fig. 3. Fig. 3 again comprises a loudspeaker-room-microphone system LRM, a loudspeaker L, a speaker position S and a microphone M. Supplementary to the LRM system shown in Fig. 2, Fig. 3 additionally illustrates a speaker signal s(n) and the pulse response h(n) of the transmission path between the loudspeaker L and the microphone M. Fig. 3 also includes the basic structure of a signal processing path for compensating for feedback, this signal processing path comprising an adaptive filter $\hat{h}(n)$ and a summing element Σ_1. As shown in Fig. 3, the adaptive filter $\hat{h}(n)$ is used in this case to generate a feedback signal $\hat{d}$ (n) from the signal x(n) for controlling the loudspeaker L. In addition, as shown in Fig. 3, the output signal $\hat{d}$ (n) from the adaptive filter $\hat{h}(n)$ is subtracted in this case from the microphone signal y(n) at the summing element Σ_1, thus generating the signal e(n) for adapting the filter coefficients of the adaptive filter $\hat{h}(n)$.

[0020] In this case, the adaptive filter

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

is used to attempt to estimate the pulse response h(n) of the transmission path between a loudspeaker L and a microphone M. Convoluting the loudspeaker signal x(n) with the estimated pulse response allows estimation of the feedback signal $\hat{d}$ (n). The aim in this case is for the estimation $\hat{h}(n)$ of the pulse response of the loudspeaker-room-microphone system to effectively match the real pulse response h(n) of the transmission path between the loudspeaker L and the microphone

M. If this is the case, the overall system can be decoupled by subtracting the estimated feedback (feedback signal $\hat{d}$ *(n)* from the microphone signal y(n).

[0021]   However, feedback compensation proves to be particularly difficult in practice since adaptation of the filter h (n) is disrupted by the great correlation between the excitation signal x(n) for the loudspeaker and the local signal s(n) from the speaker S (the speaker signal is, of course, likewise reproduced by the loudspeaker L):

$$E\{x(n)s(n+l)\} \neq 0$$

[0022]   Adaptive algorithms which converge towards the so-called Wiener solution attempt to achieve the following solution during the convergence process:

$$\hat{H}_{opt}(\Omega) = \frac{S_{xy}(\Omega)}{S_{xx}(\Omega)} = H(\Omega) + \frac{S_{xs}(\Omega)}{S_{xx}(\Omega)}$$

[0023]   In this case, the variables $S_{xy}(\Omega)$, $S_{xs}(\Omega)$ and $S_{xx}(\Omega)$ denote the cross-power density spectra between the signals x(n) and y(n) and between x(n) and s(n) and also the autopower density spectrum of the signal x(n). It should be taken into account that this does not represent the desired solution

$$\hat{H}_{opt}(\Omega) = H(\Omega)$$

[0024]   For this reason, adaptation is usually carried out only when the short-term power of the excitation signal falls (whenever the person who is speaking pauses for a short moment). During this time, the correlation between the excitation signal x(n) and the feedback component in the microphone signal is considerably larger than the correlation between the excitation signal x(n) and the otherwise prevailing local speech signal s(n).

[0025]   Furthermore, the background noise which is usually present can be replaced with artificially generated back-ground noise during pauses in speech. In this case too, the cross-correlation between the excitation signal x(n) and the local signal s(n) is considerably reduced. However, in such situations, the signal-to-noise ratio is then also very small, for which reason adaptation can be carried out only with very small step sizes. Another possible way of reducing cross-correlation is afforded by non-linearities which are inserted into the loudspeaker path. However, these non-linearities then also have an adverse effect on the reproduction of audio signals which is effected using the same loudspeaker system. If the great technical efforts made to optimize audio signal reproduction in motor vehicles are taken into account in this case, this procedure cannot be considered as a realistic way of compensating for the feedback in the passenger compartment communication systems in motor vehicles.

[0026]   Thus, a combination of all of the methods presented above is used in most contemporary systems to reduce cross-correlation. Nevertheless, during real operation, it is often possible to identify only the pulse response in those frequency ranges which have pronounced feedback. As a result of the poor matching at the remaining frequencies, feedback compensators often generate quiet but nevertheless audible artefacts which may be perceived to be unpleasant.

[0027]   There have previously been only a few systems for passenger compartment communication. All of the known examples of methods and arrangements for suppressing or compensating for feedback have the disadvantage that either the adaptation of an adaptive filter, which is used in the filtering method, is disrupted by the nature and correlation of the signals to be processed or undesirable oscillation in the attenuation response of the adaptive filters is caused, for example, by the method of operation. These and other artefacts, for example the filtering ability (which is restricted to high-level feedback) of the passive noise reduction systems which are present according to the prior art or the fact that the acoustic localization and the visual localization of a speaker do not match, constitute considerable disadvantages of the known systems.

[0028]   It is an object of the present invention to provide an arrangement and a method which exhibit improved adaptation of the filtering methods, which do not have the above-mentioned disadvantages.

SUMMARY

[0029] The object is achieved by means of the combination of active noise compensation methods with the use of psycho-acoustic effects of spatial hearing to effect of considerably higher stability of the electro-acoustic feedback loops, a reduction in artefacts and an improvement in the matching between the acoustic localization and the visual localization of a speaker.

[0030] In particular, the system according to the invention comprises a system for improving the acoustical communication between interlocutors in a room comprising at least two positions where the interlocutors are to be located in the room; at least one microphone located in the vicinity of each of said interlocutor positions in the room for generating electrical signals representative of acoustical signals present at the respective interlocutor positions; at least one loudspeaker located in the room for converting electrical signals into acoustical signals; and a signal processing unit connected to the microphone(s) and loudspeaker(s), amplifying each of the electrical signals provided by the microphones and supplying the amplified microphone signals to the at least one loudspeaker; wherein the signals from the microphones to the loudspeaker are each delayed by the signal processing unit with a delay time such that the acoustical signal arriving first at one of the interlocutor positions originates from the direction of the other interlocutor position.

[0031] The method according to the invention comprises the steps of generating electrical signals representative of acoustical signals present at the respective interlocutor positions; amplifying each of said electrical signals; and converting said amplified electrical signals into acoustical signals; wherein said electrical signals are each delayed with a delay time such that the acoustical signal arriving first at one of the interlocutor positions originates from the direction of the other interlocutor position.

BRIEF DESCRIPTION OF THE DRAWINGS

[0032] The invention can be better understood with reference to the following drawings and description. The components in the figures are not necessarily to scale, instead emphasis being placed upon illustrating the principles of the invention. Moreover, in the figures, like reference numerals designate corresponding parts. In the drawings:

Fig. 1   shows a block diagram of the arrangement of a passenger compartment communication system,

Fig. 2   shows a structure of an arrangement for suppressing feedback,

Fig. 3   shows a structure of an arrangement for compensating for feedback,

Fig. 4   shows the relationship between the loudness of different loudspeaker signals and source localization,

Fig. 5   shows a structure of a single-channel system for active feedback compensation,

Fig. 6   shows a block diagram of the complete method for suppressing feedback and improving the perception of direction, and

DETAILED DESCRIPTION

[0033] The method according to the invention described below uses a combination of active noise compensation methods and the use of psycho-acoustic effects of spatial hearing as described below.

[0034] When designing and parameterizing passenger compartment communication systems according to the invention, the psycho-acoustic effects as regards the spatial hearing sensitivities of the sound signals presented, particularly speech signals in the present case, are taken into account, in addition to the suppression of, or compensation for, feedback, in the course of communication between passengers in different seating positions in the passenger compartment of a motor vehicle. As desired, the greatest possible match between the acoustic localization and the visual localization of the respective speaker is intended to be achieved. This applies, in particular, to the rear-seat passengers since they see the front-seat passengers in front of them but the localization (which is triggered by the acoustic localization) of the front-seat passengers seems to take place behind the rear-seat passengers if the loudspeakers are situated, for example, on the parcel shelf of the passenger compartment.

[0035] Such a mismatch between different sensory impressions (in this case: visual and acoustic) may give rise to a very unnatural impression of the conversation. In reaction to such a mismatch between acoustic and visual sensory impressions, some people may also feel unwell or even nauseous. In order to avoid this, the gain of the rear loudspeakers must be limited on the basis of the temporal delay between the sounds of the loudspeaker output and the direct sound from the person who is speaking. In this case, the maximum permissible gain up to which there is still no mismatch

between the sensory impressions is described by the so-called law of the first wavefront. This psycho-acoustic effect is also referred to as the Haas effect and is described in detail, for example, in H. Haas: The Influence of a Single Echo on the Audibility of Speech, Journal of the Audio Engineering Society, Vol. 20, pages 145 - 159, March 1972.

[0036]   Fig. 4 shows the results of a psycho-acoustic investigation into directional localization and the perceived volume of speech in loudspeaker performance (see E. Meyer, G. R. Schodder: Über den Einfluss von Schallrückwürfen auf Richtungslokalisation und Lautstärke bei Sprache [The effect of sound reflection on directional localization and volume in speech], Nachrichten der Akademie der Wissenschaften in Göttingen, Math-phys. C1. 6, pages 31-42, 1952). In this case, Fig. 4 shows the results of psycho-acoustic test series in which test subjects were to adjust the perceived volume of the identical loudspeaker signals from two separate loudspeakers, which were at an equal distance from the test subject, on the basis of prescribed criteria, one of the two loudspeaker signals being reproduced with a time offset with respect to the second loudspeaker signal and this delay time between the two loudspeaker signals being additionally varied in the test series. In this case, the differences in level (in dB), which were set, on average, by the test subjects on the basis of particular prescribed criteria, between the two loudspeaker signals, which were reproduced with a time offset with respect to one another, are plotted against the delay time (in ms) in performance between these two signals.

[0037]   In this case, two loudspeakers were respectively placed at an angle of 40° and -40° in front of a test subject. Both loudspeakers reproduced the same previously recorded signal, one of the loudspeaker signals being output with a time delay of a few milliseconds (abscissa in Fig. 4). During the test, 20 test subjects were successively asked to adjust the gain of that loudspeaker which output the signal with a time delay in such a manner that

- the same loudness of the two loudspeaker signals was perceived (continuous line in Fig. 4),
- the signal from the loudspeaker with no delay could no longer be perceived (dashed line in Fig. 4), and
- the signal from the loudspeaker with a delay could no longer be perceived (dash-dotted line in Fig. 4).

[0038]   The terms volume and loudness used in this context relate to the same psycho-acoustic sensitivity variable and differ only in their units. They take account of the frequency-dependent sensitivity of human hearing. The psycho-acoustic variable loudness (see E. Zwicker and R. Feldtkeller, Das Ohr als Nachrichtenempfänger [The ear as a message receiver], S. Hirzel Verlag, Stuttgart, 1967) indicates how loud a sound event at a particular level, with a particular spectral composition and for a particular duration is perceived to be subjectively.

[0039]   In this case, the loudness is doubled when a sound is perceived to be twice as loud and thus allows different sound events to be compared with respect to the perceived volume. The unit for assessing and measuring loudness is the sone in this case. A sone is defined as the perceived volume of a sound event of 40 phons, that is to say the perceived volume of a sound event which is perceived to be as loud as a sinusoidal tone at the frequency of 1 kHz with a sound pressure level of 40 dB.

[0040]   At medium and high volumes, an increase in the volume by 10 phons results in the loudness being doubled. At low volumes, even a minor increase in volume results in the perceived loudness being doubled. In this case, the volume perceived by a person depends on the sound pressure level, the frequency spectrum and the behaviour of the sound over time.

[0041]   As can be seen in Fig. 4, it is possible, with a delay of, for example, 15 ms, to increase the volume level of the loudspeaker, which reproduces the otherwise identical signal with a time delay, by approximately 10 to 12 dB without shifting the localization of the signal in the direction of the loudspeaker which is thus louder. These results, which are taken from E. Meyer, G. R. Schodder: Über den Einfluss von Schallrückwürfen auf Richtungslokalisation und Lautstärke bei Sprache [The effect of sound reflection on directional localization and volume in speech], Nachrichten der Akademie der Wissenschaften in Göttingen, Math-phys. C1. 6, pages 31-42, 1952, in this case effectively match the conditions prevailing in passenger compartments of cars.

[0042]   If high-quality systems for improving passenger compartment communication in motor vehicles according to the present invention are not intended to adversely affect acoustic localization (that is to say are not intended to change spatial localization), the law of the first wavefront (the Haas effect described above) defines an upper limit for the maximum gain. This applies only in those cases in which this value is less than the maximum permissible gain. This is generally the case in high-quality passenger compartment communication systems in large, top of the range vehicles where the limitation of the maximum possible amplification of a signal by the Haas effect is effective more quickly than the limitation on the basis of the stability of the overall system.

[0043]   If the gain limited by the Haas effect does not suffice to distinctly improve the speech quality and the speech comprehensibility, the sound from the direction of the primary sound source must be amplified in a suitable manner (the person who is speaking at the time would have to speak louder) or additional loudspeakers which emit from the direction of the primary sound source (the person who is speaking) must be used for the perceived gain of the primary sound source. The latter case is a subject matter of the present invention in addition to the feedback suppression (described below) using active noise reduction methods.

[0044]   The first investigations into the superimposition of sound waves were carried out by Lord Rayleigh as early as

1878 (RAYLEIGH, LORD (1878): "The Theory of Sound", Vol. II, Chapter XIV, x282: "Two Sources of Like Pitch; Points of Silence; Experimental Methods", MacMillan & Co, London etc., 1st ed. 1877/78: pp. 104-106; 2nd ed. 1894/96 and Reprints (Dover, New York): pp. 116-118). On account of the complexity of the technical requirements for active noise suppression, particularly complex noise, a physically realistic approach to active noise suppression was described for the first time in 1933 (LUEG, P. (1933): "Verfahren zur Dämpfung von Schallschwingungen." [Method for attenuating sound oscillations] German Patent No. 655 508.). In this case, Lueg already described the use of electro-acoustic components to suppress noise but successful laboratory experiments in this respect were not carried out until 20 years later (OLSON, H. F. (1953): "Electronic Sound Absorber" U.S. Patent US 2,983,790 and OLSON, H. F. (1956): "Electronic Control of Noise, Vibration, and Reverberation." J. Acoust. Soc. Am. 28, 966-972). Nevertheless, on account of the range of technology needed, it was not yet possible at this time to implement actual applications.

[0045] Known methods and arrangements are intended to suppress or reduce emitted noise (ANC systems) or undesirable noise attenuate undesirable noise by generating extinction waves and superimposing them on the undesirable noise, the amplitude and frequency content of said extinction waves essentially being the same as that of the undesirable noise but their phase simultaneously being shifted through 180 degrees with respect to the undesirable noise. Ideally, this completely extinguishes the undesirable noise. This effect of reducing the sound level of noise in a desirable manner is frequently also referred to using the term destructive interference.

[0046] In the case of active noise suppression or noise compensation methods in passenger compartments of cars, the aim is to use additional loudspeakers or groups of loudspeakers to generate a so-called anti-noise field (see, for example, S. M. Kuo, D. R. Morgan: Active Noise Control Systems: Algorithms and DSP Implementations, John Wiley & Sons, New York, 1996) having the above-mentioned features. Such an approach can also be applied to the present problems of undesirable feedback in a passenger compartment communication system, as described below in Fig. 5.

[0047] Fig. 5 again comprises a loudspeaker-room-microphone system which, in the present case, is the passenger compartment of a car. For reasons of clarity, the illustration of the multiplicity of loudspeakers, which are typically present in such a passenger compartment, was again limited to a rear loudspeaker that belongs to the passenger compartment communication system and a loudspeaker L_K which is additionally fitted to the existing passenger compartment communication system, thus resulting in a single-channel system for active feedback compensation in the illustration shown in Fig. 5.

[0048] Fig. 5 also comprises the seating positions for passengers, which are known from Fig. 1 and are designated driver, front-seat passenger, rear left seating position R_L and rear right seating position R_R, as well as an exemplary microphone M from a multiplicity of microphones in the passenger compartment. Depending on the design of the car, additional seats or additional rows of seats having further seats may also be provided in this case. Fig. 5 also indicates the pulse response $h_{b1}(n)$ of the transmission path between the rear loudspeaker L_R and the microphone M and the pulse response $h_{s1}(n)$ between the additional loudspeaker L_K and the microphone M. As can be gathered from the arrows for the sound paths in Fig. 5, the reflections which arise in a passenger compartment of a car are also concomitantly included and taken into account in these pulse responses in this case.

[0049] Fig. 5 also comprises the signal processing components of the passenger compartment communication system, a filter $\hat{h}_{s1}(n)$, an adaptive filter $\hat{w}_1(n)$ and an arrangement for adapting the filter coefficients of the adaptive filter $\hat{w}_1(n)$. In this case, the signal y(n) obtained using the microphone M is processed by the signal processing components of the passenger compartment communication system and is used, in the form of the signal x(n), to control the rear loudspeaker L_R. At the same time, the microphone signal y(n) and the loudspeaker signal x(n), which has been filtered by the filter $\hat{h}_{s1}(n)$, are used to control the adaptation of the filter coefficients of the adaptive filter $\hat{w}_1(n)$. The loudspeaker signal x(n) which has been filtered by this adaptive filter $\hat{w}_1(n)$ is reproduced using the additional loudspeaker L_K in the loudspeaker-room-microphone system, that is to say in the passenger compartment of the car.

[0050] In this case, when the driver is speaking, the rear loudspeaker outputs the driver's microphone signal y(n), which has been converted into the signal x(n) by the signal processing components of the passenger compartment communication system, in order to improve the comprehensibility of the driver's speech signals for the rear-seat passengers in the rear left seating position H_L and the rear right seating position H_R. However, in this type of signal reproduction, there is also feedback to the driver's microphone M via the passenger compartment of the car. This signal transmission can be described, to a good approximation, by convoluting the signal x(n) with the pulse response $h_{b1,i}(n)$. Assuming linear time-invariant systems, the following thus results, in the frequency domain, for the feedback components of the sound signal:

$$F\left(e^{j\Omega}\right) = X\left(e^{j\Omega}\right) H_{b_1}\left(e^{j\Omega}\right)$$

[0051] The use of prefiltering by the adaptive filter $\hat{w}_{1,i}(n)$ before output using the additional loudspeaker L_K attempts

to extinguish the undesirable sound field of the feedback components at the microphone M, that is to say

$$X\left(e^{j\Omega}\right)\left(H_{b_1}\left(e^{j\Omega}\right)+\hat{W}_1\left(e^{j\Omega}\right)H_{s_1}\left(e^{j\Omega}\right)\right)=0$$

[0052] The transfer function denotes, in this case, transmission from the additional loudspeaker L_K to the driver's microphone via the passenger compartment of the vehicle. As can be discerned from the equation above, an adaptation method must be used to attempt to set the coefficients of the adaptive filter $\hat{w}_{1,i}(n)$ in such a manner that:

$$W_1\left(e^{j\Omega}\right)=-\frac{H_{b_1}\left(e^{j\Omega}\right)}{H_{s_1}\left(e^{j\Omega}\right)}$$

[0053] In this case, virtually all common methods, for example the NLMS algorithm, affine projection methods or the RLS method, may be used as adaptation methods (also see, in this respect, S. Haykin: Adaptive Filter Theory, 4th edition, Prentice Hall, Englewood Cliffs, New Jersey, 2002). The transfer function H_{s1} ($e^{j\Omega}$) in the denominator of the above equation proves to be problematic in this case in the real application of the method. Should the z transform of this pulse response have zeros outside the unit circle or in the unit circle, the optimal solution according to

$$X\left(e^{j\Omega}\right)\left(H_{b_1}\left(e^{j\Omega}\right)+\hat{W}_1\left(e^{j\Omega}\right)H_{s_1}\left(e^{j\Omega}\right)\right)=0$$

represents an unstable filter. In order to avoid this, the so-called filtered xLMS algorithm is frequently used. In this case, a previously filtered variant rather than the input signal x(n), that is to say the loudspeaker signal from the rear loudspeaker L_K itself, is used to calculate the filter correction (adaptation of the filter coefficients). In this case, prefiltering should ideally be carried out with the pulse response

$$\hat{h}_{s_1,i}(n)=h_{s_1,i}(n)$$

[0054] For further details on active noise suppression methods, reference is made to S. M. Kuo, D. R. Morgan: Active Noise Control Systems: Algorithms and DSP Implementations, John Wiley & Sons, New York, 1996.
[0055] In addition to feedback suppression, an active arrangement, as illustrated in Fig. 5, has yet further advantages for improving comprehensibility in passenger compartments of vehicles:

- Outputting speech signals from the driver using the additional side loudspeaker L_K, which is positioned in the vicinity of the front-seat passenger, also improves comprehensibility for the front-seat passenger.

- The front-seat passenger loudspeaker L_K additionally means, for the rear-seat passengers, a sound source which likewise emits signals from the front. This increases the primary wavefront for the Lombard effect (change in the voice in loud surroundings), and greater amplification of the sound signals is possible (while simultaneously retaining the correct acoustic perception of direction).

- If the driver's microphone is situated in the vicinity of the driver, the sound which is added in phase opposition and is intended to extinguish the undesirable sound components - at least at low frequencies - also improves the driver's perception of echoes.

[0056] The advantages of the two methods described are combined below in such a manner that the greatest possible improvement can be achieved overall. In this case, it should be taken into consideration that the results obtained and described here may also be applied to the opposite conditions, that is to say when the front-seat passenger is speaking and the remaining passengers are listening.

[0057] The two effects and method approaches previously described may be combined in this case, according to the invention, in such a manner that it is possible to achieve both greater amplification of the desired sound signals (without violating the law of the first wavefront) and active suppression or compensation of acoustic feedback in an arrangement. Fig. 6 shows the arrangement (which is used for this purpose) of the inventive combination of methods, which is based on the structure of the arrangement shown in Fig. 5.

[0058] In this case, Fig. 6 again comprises a loudspeaker-room-microphone system which, in the present case, is the passenger compartment of a car. Fig. 6 also comprises the seating positions for passengers, which are known from Fig. 1 and Fig. 5 and are designated driver, front-seat passenger, rear left seating position R_L and rear right seating position R_R, as well as an exemplary microphone M from a multiplicity of microphones in the passenger compartment. Fig. 6 also indicates, in the LRM system, the pulse response $h_{s1}(n)$ of the transmission path between a loudspeaker L_{K1} on the front-seat passenger's side and a microphone M and the pulse response $h_{s2}(n)$ between a loudspeaker L_{K2} on the driver's side and the microphone M.

[0059] Fig. 6 also comprises the additional signal processing components of the passenger compartment communication system, a first filter $\hat{h}_{s1}(n)$, a first adaptive filter $\hat{w}_1(n)$, a second filter $\hat{h}_{s2}(n)$, a second adaptive filter $\hat{w}_2(n)$ and a respective arrangement for adapting the filter coefficients of the adaptive filters $\hat{w}_1(n)$ and $\hat{w}_2(n)$. In this case, the signal y(n) obtained using the microphone M is processed by the signal processing components of the passenger compartment communication system and is used, in the form of the signal x(n), to directly control the left-hand and righthand loudspeakers (not described in any more detail) in the rear part of the passenger compartment (rear seat). In addition, the microphone signal y(n) and the loudspeaker signal x(n), which has been filtered by the first filter $\hat{h}_{s1}(n)$, are again used to control the adaptation of the filter coefficients of the first adaptive filter $\hat{w}_1(n)$. The loudspeaker signal x(n) which has been filtered by this first adaptive filter $\hat{w}_1(n)$ is reproduced using the loudspeaker L_{K1} in the loudspeaker-room-microphone system, that is to say in the passenger compartment on the front-seat passenger's side of the car. In addition, as shown in Fig. 6, the microphone signal y(n) and the loudspeaker signal x(n), which has been filtered by the second filter $\hat{h}_{s2}(n)$, are used to control the adaptation of the filter coefficients of the second adaptive filter $\hat{w}_2(n)$. The loudspeaker signal x(n) which has been filtered by this second adaptive filter $\hat{w}_2(n)$ is reproduced using the loudspeaker L_{K2} in the loudspeaker-room-microphone system, that is to say in the passenger compartment on the driver's side of the car.

[0060] In addition to the loudspeaker L_{K1} on the front-seat passenger's side, the loudspeaker L_{K2} which is usually fitted in the driver's door on the driver's side is also additionally used, according to the embodiment of the method according to the invention shown in Fig. 6, to improve localization and to improve active feedback compensation. The use of this loudspeaker affords an additional sound source in the immediate vicinity of the speaker (the driver in the present example). As regards the Haas effect described further above, this means that the primary sound source of the speech signal in the passenger compartment can be additionally amplified, and an even greater resultant gain is thus possible, without changing the impression of the direction, that is to say the localization. However, when setting the adaptive filters, it must be taken into account in the present case that a plurality of anti-noise loudspeakers and channels are now used. This mainly makes it necessary to commonly standardize the adaptation step size (for details of this see S. M. Kuo, D. R. Morgan: Active Noise Control Systems: Algorithms and DSP Implementations, John Wiley & Sons, New York, 1996).

[0061] However, the additional loudspeaker in the vicinity of the speaker cannot be used in this case as in conventional active noise compensation applications since the person who is speaking would otherwise perceive their own speech signal as a clear echo. For this reason, the magnitude of the transfer function $W_2(e^{j\Omega})$ must be limited to a value which prevents the perception of one's own speech signal which arrives after a time delay. The same applies to outputting the speaker's signal on the front-seat passenger's side but the upper limit may be selected in this case to be larger than on the speaker's side (the distance between the loudspeaker L_{K1} on the front-seat passenger's side and the speaker on the driver's side is considerably larger than the corresponding distance between the loudspeaker L_{K2} on the driver's side and the speaker who is the driver in the present example).

[0062] Since echoes are perceived to be considerably less disruptive at low frequencies and a longer delay time before such echoes arrive is tolerated and, in addition, the performance of active noise and feedback compensation methods is considerably better at low frequencies, it is desirable to restrict the signals which have been reproduced to their low-frequency signal components on that side of the passenger compartment which is in the vicinity of the speaker. For this reason, low-pass filters are respectively integrated in the signal output or adaptation path in the vicinity of the speaker, as shown in Fig. 6. The selection of the cut-off frequency of such low-pass filters depends on the geometry of the passenger compartment of the car and, in particular, on the distance between the loudspeakers and the ears of the person who is speaking and on the distance between the microphones and the ears of the person who is speaking and on the associated sound propagation times.

[0063]   In this case, the pulse responses $\hat{h}_{s1,i}(n)$ and $\hat{h}_{s2,i}(n)$ needed for signal prefiltering may either already be measured in advance or may be adaptively determined during use of the method according to the invention. The last-mentioned variant is to be preferred in this case since the seating positions or the number of passengers, for example, are unknown in advance. Since ambiguity arises when directly identifying the pulse responses using the output signals from the passenger compartment communication system (for details see E. Hänsler, G. Schmidt: Acoustic Echo and Noise Control, John Wiley & Sons, New York, 2004), it is advantageous to use the pulse responses which are estimated, for example, when compensating for radio signals. Such a method is described, for example, in G. Schmidt, T. Haulick, H. Lenhardt: Enthallung der Wiedergabe von Audiosignalen in Fahrzeugen mit Insassenkommunikationsanlagen [Der-everberating the reproduction of audio signals in vehicles having passenger communication systems], notification of invention P05051, January 2005.

[0064]   Finally, reference shall also be made to the possibility of using not only individual loudspeakers but arrays of loudspeakers. In this case, a double loudspeaker in the driver's door, for example, could be controlled using suitable prefiltering in such a manner that emission in the direction of the driver is as low as possible but maximum emitted power and thus maximum compensation for the undesirable signal components are achieved in the direction of the recording microphone.

[0065]   The advantageous effect of the invention results from the use of noise compensation methods which are active, for example, but not limited to ANC (Active Noise Cancellation) methods, thus resulting in increased stability of the method when reducing undesirable feedback and, overall, in an increase in the maximum possible reproduction level.

[0066]   Further advantages may also result if, as a result of the use of psycho-acoustic effects in the type and distribution of signal reproduction using the loudspeakers of a passenger compartment communication system, matching between the visual localization and the acoustic localization of a speaker is improved.

[0067]   Yet further advantages may also result if, as a result of the appropriate deliberate and additional use of individual loudspeakers, for example a side loudspeaker, the comprehensibility of speech signals is enhanced, for example for a front-seat passenger.

[0068]   Yet further advantages may likewise also result if, as a result of active noise compensation, the perception of echoes is also improved.

[0069]   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.   System for improving the acoustical communication between interlocutors in a room comprising
at least two positions where the interlocutors are to be located in the room;
at least one microphone located in the vicinity of each of said interlocutor positions in the room for generating electrical signals representative of acoustical signals present at the respective interlocutor positions;
at least one loudspeaker located in the room for converting electrical signals into acoustical signals; and
a signal processing unit connected to the microphone(s) and loudspeaker(s), amplifying each of the electrical signals provided by the microphones and supplying the amplified microphone signals to the at least one loudspeaker;
wherein the signals from the microphones to the loudspeaker are each delayed by the signal processing unit with a delay time such that the acoustical signal arriving first at one of the interlocutor positions originates from the direction of the other interlocutor position.

2.   The system of claim 1 wherein, in the signal processing unit, the amplification of the respective microphone signal is limited such that the level of signals not originating from the direction of the other interlocutor position exceeds the level of signals originating from the direction of the other interlocutor position by less than a given level difference.

3.   The system of claims 1 wherein at least two loudspeakers are arranged in the room; said signal processing unit amplifying and delaying each of the electrical signals provided by the microphones and supplying the amplified and delayed microphone signals to each of said loudspeakers such that the acoustical signal arriving first at one of the interlocutor positions originates from the direction of the other interlocutor position.

4.   The system of claim 3 wherein, in the signal processing unit, the amplification of the respective microphone signal is limited for each of the loudspeakers separately such that the level of signals not originating from the direction of the other interlocutor position exceeds the level of signals originating from the direction of the other interlocutor

position by less than a given level difference.

5. The system of claim 2 or 4 wherein said given level difference is depending on said delay time.

6. The system of claims 1-5 further comprising at least one additional loudspeaker supplied with a noise cancellation signal from a noise processor unit; said noise cancellation signal representing the phase-inverted noise signal in the vicinity of said microphone.

7. The system of claim 6 wherein the at least one additional loudspeaker is arranged perpendicular to the main axis of the microphone or at least one of the microphones.

8. The system of claim 6 or 7 wherein at least one of the additional loudspeakers is arranged in the vicinity of at least one of the interlocutor positions.

9. The system of claim 6, 7, or 8 wherein the noise processor unit is an adaptive filter supplied with signals from the at least one microphone and the at least one loudspeaker and generating the noise cancellation signal by extracting the noise signal in the vicinity of said microphone and inverting the phase.

10. The system of claim 9 wherein said adaptive filter uses the NLMS algorithm, affine projection methods, the RLS method or the filtered xLMS algorithm.

11. The system of one of claims 6-10 wherein the noise processor unit comprises transfer function, the magnitude of which is limited to a given value.

12. The system of one of claims 6-11 wherein the noise processor unit comprises a low pass filter unit in the signal path between the one of said microphones and the one of said loudspeakers.

13. Method for improving the acoustical communication between interlocutors in at least two positions in a room, said method comprising the steps of:

    generating electrical signals representative of acoustical signals present at the respective interlocutor positions;
    amplifying each of said electrical signals; and
    converting said amplified electrical signals into acoustical signals;
    wherein said electrical signals are each delayed with a delay time such that the acoustical signal arriving first at one of the interlocutor positions originates from the direction of the other interlocutor position.

14. The method of claim 13 wherein the amplification of the respective electrical signal is limited such that the level of signals not originating from the direction of the other interlocutor position exceeds the level of signals originating from the direction of the other interlocutor position by less than a given level difference.

15. The method of claims 13 wherein the acoustical signals converted from said amplified and delayed electrical signals are radiated in at least two positions in the room; said amplifying and delaying step is applied to each of the electrical signals generated; and the amplified and delayed electrical signals are radiated at each radiating position such that the acoustical signal arriving first at one of the interlocutor positions originates from the direction of the other interlocutor position.

16. The method of claim 15 wherein the amplification of the respective electrical signals representative of acoustical signals present at the respective interlocutor positions is limited for each of the radiating position separately such that the level of signals not originating from the direction of the other interlocutor position exceeds the level of signals originating from the direction of the other interlocutor position by less than a given level difference.

17. The method of claim 14 or 16 wherein said given level difference is depending on said delay time.

18. The method of claims 13-17 wherein at least one additional radiating position is arranged in the room; said method further comprising the step of radiating at the additional position a noise cancellation signal; said noise cancellation signal representing the phase-inverted noise signal in the vicinity of the respective interlocutor position.

19. The method of claim 18 wherein the at least one additional radiating position is arranged perpendicular to the main

axis of the position or at least one of the position where the electrical signal representative of acoustical signals present at the respective interlocutor positions is picked up.

20. The method of claim 18 or 19 wherein at least one of the additional radiating positions is arranged in the vicinity of at least one of the interlocutor positions.

21. The method of claim 18, 19, or 20 further comprising the steps of:

   adaptive filtering of signals from the at least one microphone and the at least one loudspeaker and generating the noise cancellation signal by extracting the noise signal in the vicinity of said interlocutor positions and inverting the phase.

22. The method of claim 21 wherein said adaptive filtering is according to the NLMS algorithm, affine projection methods, the RLS method or the filtered xLMS algorithm.

23. The method of one of claims 18-22 wherein the adaptive filtering includes a transfer function, the magnitude of which is limited to a given value.

24. The method of one of claims 18-23 further comprising the step of low pass filtering in the signal path between the one of said positions where the signals relating to the interlocutor positions are picked up and the one of said radiating positions.

Fig. 1

Fig. 2

Fig. 3

Differences in level between the first and second wavefronts (angular difference = 80°)

- — — First wavefront cannot be perceived
- ——— Same loudness of the two wavefronts
- —·— Second wavefront cannot be perceived

Fig. 4

Adaptation

$\widehat{h}_{S_1}(n)$

$\widehat{n}_1(n)$ $x(n)$

LRM

Passenger compartment of the car

R_R

L_R

$x(n)$

Front-seat passenger

$h_{S_1}(n)$

Driver

$y(n)$

$h_{L_1}(n)$

M

R_L

Signal processing components
of the passenger compartment
communication system

Fig. 5

Lk1

Passenger compartment
of the car

R_R

$h_{S_1}(n)$

Front-seat passenger

LRM

$y(n)$

M

Driver

R_L

$h_{S_2}(n)$

Lk2

Low-pass filter

$\hat{w}_2(n)$

Low-pass filter   Adaptation

$\hat{h}_{S_2}(n)$

$\hat{w}_1(n)$

Adaptation

$\hat{h}_{s_1}(n)$

$y(n)$ $x(n)$

Further signal processing components
of the passenger compartment
communication system

Fig. 6

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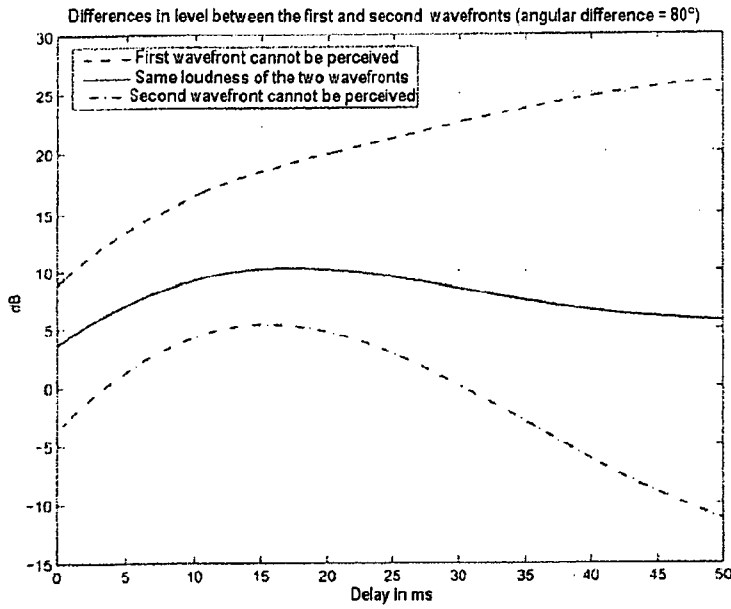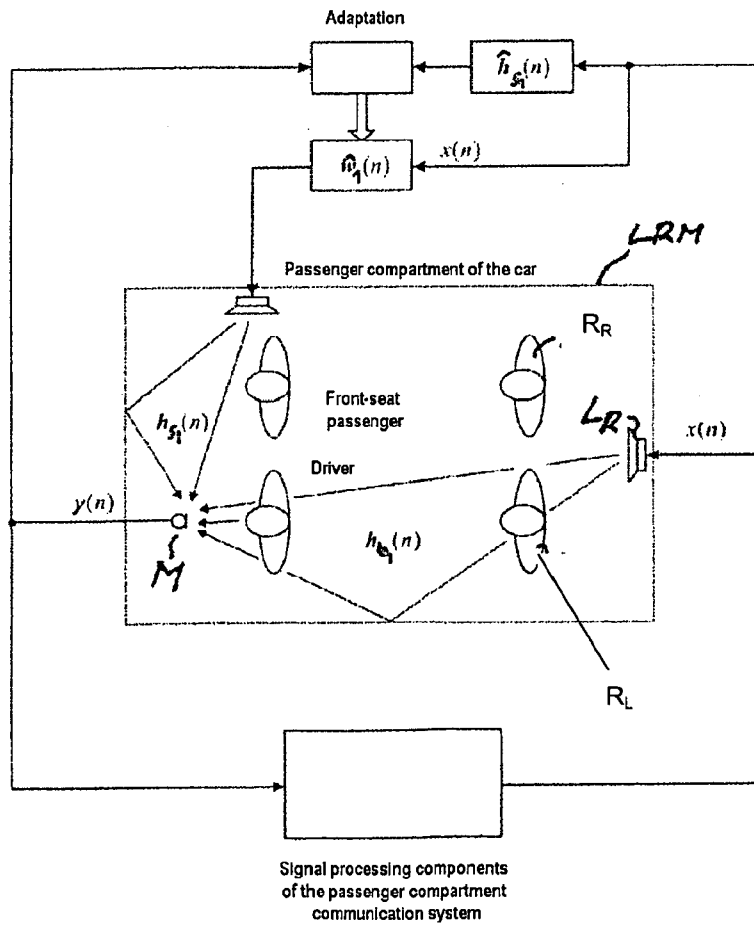
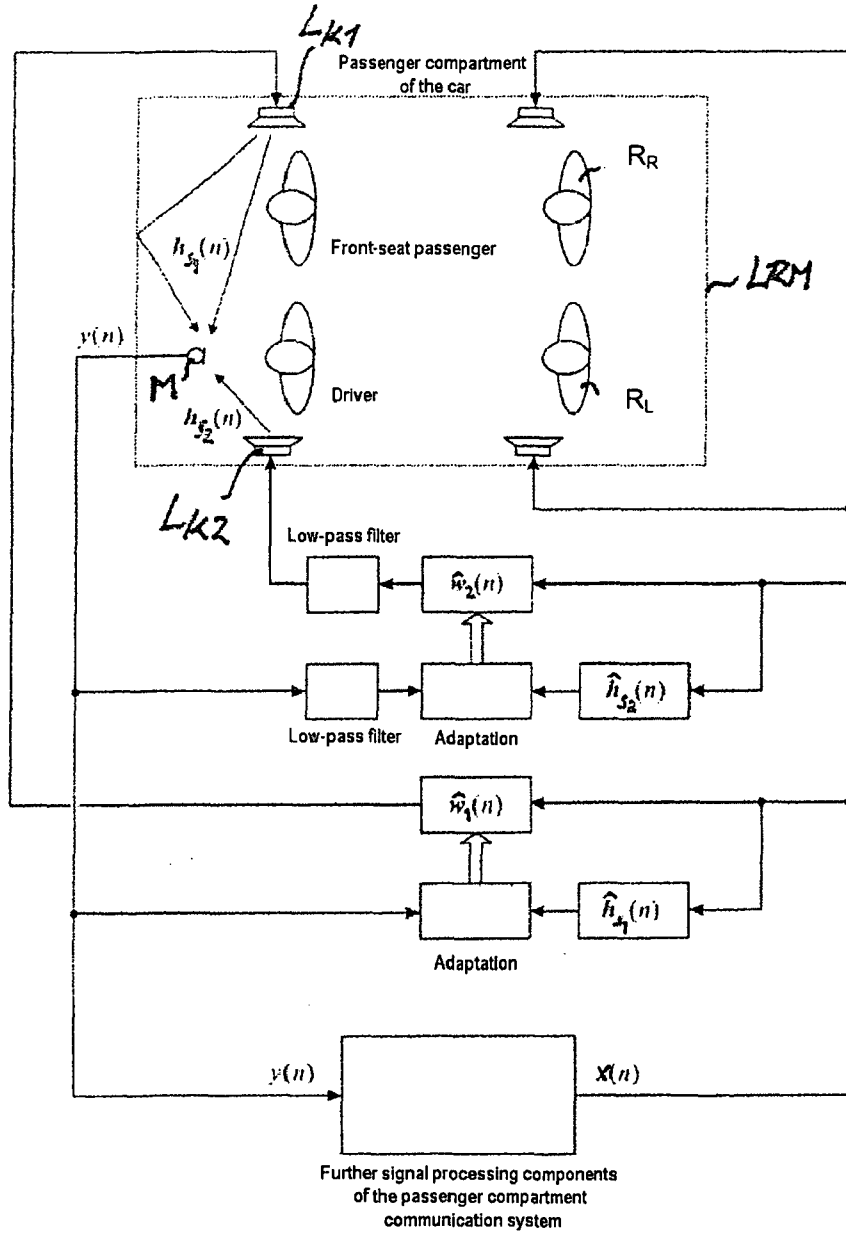## EUROPEAN SEARCH REPORT

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