(54) Title: SYSTEM FOR NOISE SUPPRESSION, TRANSCEIVER AND METHOD FOR NOISE SUPPRESSION

(57) Abstract: A system with an accompanying method is provided to improve the signal-to-noise ratio (SNR) of noisy speech by suppressing acoustic background noise. The background noise consists of narrow band noise from rotating machine, audio signals from stereo-loudspeakers of audio entertainment device, and other ambient noise. In this system/apparatus, a microphone senses the speech intermingled with the background noise, and another microphone senses the noisy background. In addition, a measurement sensor is used to measure RPM (revolutions-per-minutes) of the rotating machine and two wires are used to acquire audio signals from the stereo-loudspeakers of the audio entertainment device. Furthermore, to provide better suppression performance for the acoustic audio signals, the characteristics of these loudspeakers are used to compensate for the distortion caused by the loudspeakers. Adaptive comb filters and adaptive FIR filters are applied to estimate the ambient noise and suppress the background noise. After processing, the system outputs the enhanced speech signal with higher SNR.
Published:
— with international search report

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SYSTEM FOR NOISE SUPPRESSION, TRANSCEIVER AND METHOD FOR NOISE SUPPRESSION

TECHNICAL BACKGROUND AND PRIOR ART

This invention relates to a system for the suppression of noise, an accompanying method or a transceiver.

In many cases, noise corrupts a speech signal and hence significantly degrades the quality of recognition of the speech signal. An example for such noise is background noise intermingled with the speech signal acquired by a microphone, a hand-free phone, a handset or the like.

It is important to recognize speech in a noisy environment, e.g. a night club, a sport club, a Karaoke room, a hands-free communication system in a vehicle, especially a car, a helicopter, a tank or the like. Furthermore, noise suppression is useful in a live reporting system, a public addressing system or the like.

The recognition of speech or voice can be done by an automatic speech recognition system or by at least one human listener.

The undesirable background noise can be of different sources. For example, making telephone calls out of a driving car, the driving noise, especially the noise of the engine, is a dynamically varying kind of noise that results in poor recognition of the speech, particularly in a hands-free speaking environment of the car. The addressee permanently hears a contaminated acoustic signal, in which the voice of the driver is included but difficult to understand. As a consequence, the driver has to speak up or take the handset of the telephone, which binds his attention to the handset.
and not the traffic - a very undesirable effect. Another scenario relates to signals from an audio system that worsens the recognition of the speech intermingled with the audio noise.

Moreover, there are lots of sites which need better recognition of speech and/or better understanding because of a noisy background. Some sites, additional to the above mentioned scenarios, are: airplanes, helicopters, airports, trains, buses, train stations, bus stops, construction sites, highways, streets or the like.

In [1] a concept and basic approach for adaptive noise cancellation are given. It can be used to eliminate background noise and improve a signal-to-noise-ratio (SNR). Therefore, a primary input containing a corrupted signal and a reference input containing noise correlated in some unknown way with the primary noise are used. This reference input is adaptively filtered and subtracted from the primary input to obtain the signal estimate. Adaptive filtering before subtraction allows the treatment of inputs that are deterministic or stochastic, stationary or time variable. Wiener solutions are developed to describe asymptotic adaptive performance and output SNR for stationary stochastic inputs, including single and multiple reference inputs. These solutions show that, when the reference input is free of signal and certain other conditions are met, noise in the primary input can be essentially eliminated without signal distortion. Further, it is shown that in treating periodic interference, the adaptive noise canceler acts as a notch filter with narrow bandwidth, infinite null, and the capability of tracking the exact frequency of the interference; in this case, the canceler behaves as a linear, time-invariant system, with the adaptive filter converging on a dynamic rather than a static solution.
In [2] a voice operated switch in a noisy environment is described. This switch is capable of distinguishing between voice and non-voice (noise).

In [3] an approach to improve the basic idea of [1] to eliminate cross-talk effects between noise and speech signals is presented.

In [4] an adaptive noise suppressing device is introduced. Here, the characteristics of an adaptive filter are adjusted automatically dependent on variations of the input signal.

In [5] a system utilizing two specially-built microphones that have good near field response and poor far field response to produce signals with noise components having high correlations is disclosed.

Document [6] uses a filter bank for band-dividing the input signal from the main microphone and the second noise component from the reference microphone, and a noise cancelling circuit for obtaining a phase difference between the input signal and the second noise component with respect to each divided band of the filter bank so as to correct the input signal based on the phase difference and for cancelling the first noise component in the input signal by use of the corrected input signal.

In [7] Hunt adopts an adaptive filtering technique which is employed using the power spectra in both channels, i.e. in a speech channel and in a reference channel, when speech is not present in the speech channel to obtain a relationship between the environmental noise power spectra in the two channels. When speech is present in the speech channel, a prediction of the environmental noise power spectrum on that channel is obtained from the power spectrum of the noise on the reference channel and the relationship between the noise power spectra on the two channels previously obtained.
In [8] a method to adjust the updating step size of the adaptive filter is proposed so that the system has a better tracking ability while the desired speech does not exist, and otherwise has a smaller residual noise while the expected speech appears.

All of the above cited documents face the disadvantage that some kind of noise, e.g. noise of some sort of machine or noise of a loudspeaker, is not considered in an appropriate and favourable way.

**SUMMARY OF THE INVENTION**

It is an object of the present invention to provide an acoustic noise reduction system and/or apparatus to be able to cancel narrow band and broadband noises simultaneously.

More specifically, it is an object of the invention to reduce background noise to an acceptable level even when the signal-to-noise-ratio (SNR) is low.

Another object is to provide a noise reduction system and/or apparatus which reduces audio signals from an audio source. Such an audio source drives at least one loudspeaker and is used as entertainment device in a car, in a club, at home or the like.

Yet another object of the invention is to remove the interference to the signal of the reference microphone caused by narrow band noise and audio signals.

The objects of the present invention are achieved by the features of the independent claims. Additional features result from the dependent claims.
The present invention comprises a system for noise suppression of a speech signal that is intermingled with general noise. The system comprises an Input Unit for receiving the speech signal intermingled with the general noise. The Input Unit further comprises a Sound Processing Unit and a sensor for measuring a rotating rate of a device, in the course of which the Sound Processing Unit generates a first noise signal of the device using the value of the sensor. This noise signal stands for a reference signal for the noise of the device because of its operation. This reference signal - evaluated from the rotation of the device - is later used to suppress the real noise that emerges from this device.

Furthermore, the system comprises a Processing Unit with a first adaptive filter to evaluate a dynamically varying second noise signal out of the first noise signal and with a first calculation means to evaluate a first noise suppressed signal out of the second noise signal and the speech signal that is intermingled with the general noise.

It is an advantage of this system to suppress a noise signal that emerges from the rotation of a device, said device is e.g. a rotating device/machine, particularly an engine. Such an engine produces noise dependent from its revolution per time, the noise becomes sharper, particularly the frequency of the noise gets higher, when the revolutions increase. Hence the noise is directly correlated to these revolutions and measuring the revolutions, e.g. by a revolution counter, allows to determine the frequency of the noise of the engine.

From the known revolutions per time unit (e.g. minute) it is possible to generate a noise with the aid of a Sound Processing Unit, e.g. a sine-wave generator. This generated noise is used to reduce the general noise of the speech signal that is intermingled with said general noise.
The speech signal intermingled with the general noise can be acquired by a microphone.

It is an embodiment of the present invention to realize said adaptive filter as an adaptive comb filter that is used to suppress the narrow band noise. Particularly, this adaptive filter can be switched on, if noise from the rotating device exists and otherwise, it can be switched off and no noise output from the Sound Processing Unit will be provided in this case.

Moreover, objects of the invention are achieved by a system for noise suppression (out) of a speech signal that is intermingled with general noise. This system comprises an Input Unit for receiving the speech signal intermingled with the general noise and at least one audio signal from an audio source, e.g. a mono or a stereo device for entertainment or the like. Furthermore, the system comprises a Processing Unit with a second adaptive filter to evaluate a third noise signal out of the audio signal and with a second calculation means to evaluate a second noise suppressed signal out of the third noise signal and the speech signal.

It is an advantage of this system, that noise coming from an audio source can be suppressed with the aid of the signal from this audio source, e.g. the signal that is sent to at least one loudspeaker connected to the audio source. Furthermore, the transfer function of this at least one loudspeaker is used to suppress the noise that the audio source produces. Here, the sound from the audio source, entertainment as music or speech, is regarded as noise that has to be suppressed in order to understand the real speech signal (intermingled with this noise) properly. This transfer function of the at least one loudspeaker can be used in a compensator unit to compensate the distortion of this at least one loudspeaker.
It should be noted that more than one loudspeaker can be provided in course of which each loudspeaker might have a different distortion and hence a separate compensation unit has to be provided. This is important for stereo audio systems, quadraphonic sound or the like. For each audio channel an adaptive filtering can be provided.

It is an embodiment that the transfer function of each loudspeaker can be calculated offline.

Furthermore, objects of the invention are achieved by a system that embodies both features described above: first the suppression mechanism of the noise from the device (e.g. rotating device, engine or motor) and second the suppression mechanism of the noise of the audio signal.

It is advantageous for suppression of noise signals in an efficient way to provide a calculation system that evaluates the final noise suppressed signal in stages, first considering the noise from the device and second considering the noise of the audio signal or vice versa. It should be noted that if some kind of noise does not exist, it should not be considered in calculation. This can be achieved by a switching mechanism of the adaptive filters (on or off) dependent on the existence of the respective noise signal.

An example for implementation is a flag-mechanism: If a respective noise does not exist or is below a predefined level, the respective adaptive filter will be switched off and it won't be considered in the respective calculation unit.

Furthermore, it is an embodiment of the invention to provide an Ambient Noise Estimator within the Processing Unit. The Ambient Noise Estimator takes a background noise signal (just the background noise not the speech signal that has to be identified) into consideration. This background noise can be
recorded or received by a microphone. Within the Ambient Noise Estimator a calculation takes place to suppress noise from the (rotating) device and the audio source within the background noise signal. Thus, this modified background noise signal is an estimation of the ambient noise.

As stated above, although within the Ambient Noise Estimator a switching mechanism exists calculation of the ambient noise is done dependent on the existence of the noise from the (rotating) device and/or the audio signal. This switching of the adaptive filters (see also as described above) can be done within a noise/audio signal detector wherein flags are set dependent on the existence of different kinds of noise.

It has to be emphasized that the adaptive filters can be FIR-filters and some of them can be comb-filters also.

Moreover, a voice detection unit can be provided to switch the adaptive filters within the Processing Unit dependent on the existence of the speech signal (intermingled with the general noise). Particularly, if this speech signal is below a predefined level, it is considered to be non-existent and therefore no calculation to suppress the noise within this speech signal needs to be done.

It is a result of the above described system(s) that an output signal is provided that has a better signal-to-noise-ratio than the speech signal (intermingled with general noise).

Further, the signal processing in the described system(s) is preferably done on a digital signal. Hence a conversion from analogue to digital can be done within the Input Unit. The signals acquired of the microphones are converted into digital signals as well as the analogue signals of the audio source. The generated signal of the (rotating) device can be calculated directly as a digital signal by the sine-wave
generator. To achieve the object of a speech signal with suppressed noise, the digitally processed signals have to be transformed into an analogue output signal that is presented as a result of the invention.

Processing of this noise suppressed signal that can be of digital or analogue type - can be done. One possibility of processing the digital output signal is to do an automatic speech recognition. An object of such speech recognition can be a controlling of some kind of function, e.g. voice detection, recognition and control of some functions in a car while driving. Another possibility is the analogue presentation of the converted speech signal to a human listener who will be able to understand what the speaker said despite ambient noise of different types.

It is another embodiment of the present invention that the described system is a transceiver.

It is yet another embodiment of the present invention to provide a method for noise suppression to be executed on any of the above described systems.

**BRIEF DESCRIPTION OF THE DRAWINGS**

Examples of the present invention will be described in detail in view of the following drawings.

Fig. 1 shows a block diagram of the main units to achieve the goal of noise suppression;

Fig. 2 shows the detailed illustration of the Input Unit of Fig. 1;

Fig. 3 shows the detail of the Processing Unit of Fig. 1;

Fig. 4 shows the detail of the Ambient Noise Estimator of Fig. 3.
DETAILED DESCRIPTION OF THE DRAWINGS

In Fig.1 a system composed of an Input Unit 101, a Processing Unit 102 and an Output Unit 103 is shown. The operations of the system for noise suppression and the function of each respective unit 101, 102 and 103 can be summarized as follows.

First, sensors in the Input Unit acquire signals that are processed by the system. These signals are: the speech signal intermingled with the general noise and various kinds of other signals embodying the background noise. Then those signals are, if necessary, A/D converted (see Fig.2 for details) and input to the Processing Unit 102.

Second, the Processing Unit 102 is used to suppress the background noise of various kinds. The Processing Unit 102 can be divided into modules, i.e. an Ambient Noise Estimator 104 and a Noise Reduction Module 105. As first part of Processing Unit 102, the Ambient Noise Estimator 104 estimates the ambient noise except the noise from a (rotating) device, e.g. a rotating machine or an engine, and audio signals from an audio system, e.g. an audio entertainment device. The signals from the Input Unit 101 along with the estimated ambient noise are processed by the Noise Reduction Module 105. Finally the enhanced speech signal is converted to an analogue signal by a D/A converter (see 32 in Fig.3) and output through the Output Unit 103.

As shown in Fig.2, Fig.3 and Fig.4, it should be emphasized that the system possesses several unique characteristics.

- First, the transfer functions of the loudspeakers (see 10 and 11 in Fig.2) are taken into account to provide better presentation of the acoustic stereo audio signals from an audio source, e.g. an audio entertainment device, as it can compensate the distortion caused by the loudspeakers. These audio signals contribute to the background noise.
Second, as shown in Fig.3 and Fig.4, the background noise $bn(k)$ acquired by the reference microphone 19 contains narrow band noise from the (rotating) device, e.g. a rotating machine, acoustic audio signals from the audio source and other ambient noise. In this system, the estimated narrow band noise and audio acoustic signals are subtracted from the background noise $bn(k)$, as a result the estimated ambient noise $er2(k)$ is obtained. This signal $er2(k)$ can be used as a reference signal for cancelling the ambient noise.

Third, the narrow-band Noise/Audio Signal Detector 26 is used to control operations of both the Ambient Noise Estimator 104 and Noise Reduction Module 105.

Other objects, features and advantages according to the present invention will be presented in the following detailed description of the illustrated embodiments when read in conjunction with the accompanying drawings.

In the previous part, the functions and operations of the system shown in Fig.1 have been described. Now, supplementary information is provided to show the content and functionality of each unit named in Fig.1.

In Fig.2 the Input Unit 101 is shown, said Input Unit 101 comprising: a microphone 1, a reference microphone 19, a sensor 5 for measuring revolutions per minute (RPM) of the rotating machine, two wires 12 and 13 for acquiring a stereo audio signal from an Audio Entertainment Device 16, A/D converters 3, 21, 14 and 15, pre-amplifiers 2 and 20, loudspeaker compensators 17 and 18, and a sine-wave generator 4.

Fig.3 shows the Processing Unit 102, comprising: an adaptive comb filter 8, adaptive filters 28 and 30, and calculations units 9, 29 and 31, further referred to as adders. The Output
Unit 103 includes a D/A converter 32 and an Output Connection Unit 33.

**Fig. 2** shows detailed illustration of the Input Unit 101. The desired microphone 1 acquires the speech signal which is speech intermingled with general background noise. After amplified by the pre-amplifier 2, this signal is A/D converted to a digital desired signal $d(k)$ by using the A/D converter 3.

The reference microphone 19 senses the background noise, which contains narrow-band noise from rotating machine, acoustic audio signals from audio entertainment device, and other ambient noise. This reference signal is amplified by the pre-amplifier 20 and A/D converted to a digital signal $b_n(k)$ by using A/D converter 21.

The sensor 5, which can be a tachometer, an accelerometer or the like, measures revolutions per minute (RPM) of the (rotating) device, further referred to as rotating machine. The RPM is used to compute a fundamental frequency $f_0$ of this narrow-band noise. This fundamental frequency $f_0$ is used to excite a sine-wave generator 4 to generate digitised sine and cosine waves with the frequency $f_0$ and its harmonic frequencies. The sine and cosine waves are labelled $s_i(k)$ and $c_i(k)$, $(i=1,\ldots,M)$, respectively, where $M$ is the total number of the frequency components.

The signals from the audio entertainment device 16 are used to drive both loudspeakers 10 and 11 to generate the acoustic stereo audio signals. The wires 12 and 13 contain these stereo signals which are converted into digital signals using the A/D converters 14 and 15. These digitised signals are labelled $l(k)$ and $r(k)$, which represent the signals from the left channel and from the right channel, respectively. The left loudspeaker compensator 17 and right loudspeaker compensator 18 are used to compensate the distortion of the
loudspeakers to provide better presentation of the acoustic stereo audio signals. The compensated signals are labelled rl(k) and rr(k) (second letter "l" for "left", "r" for "right").

Fig.3 shows the Processing Unit 102 (see also Fig.1). The adaptive comb filter 8 and two adaptive FIR filters 28 and 30 are used to reduce the background noise. The Ambient Noise Estimator 301 (see also Fig.4) provides a reference signal for reducing ambient noise. Two detectors, i.e. a Narrow-band Noise/Audio Signal Detector 26 (called noise detector hereafter) and a Voice Detector 27 control operation of other components within the system. Before explaining the operation of this Processing Unit 102, the control signal from both detectors 26 and 27 is defined as follows:

- $\text{flag0} = 0$ when narrow-band noise does not exist, and $\text{flag0} = 1$ when it exists.
- $\text{flag1} = 0$ when stereo audio signals do not exist, and $\text{flag1} = 1$ when they exist.
- $\text{flag2} = 0$ when desired voice (speech signal) does not exist, and $\text{flag2} = 1$ when it exists.

These flags incorporate a switching mechanism dependent on the state of each flag. If some kind of noise does not exist or has a signal strength that is below a predefined level, this noise has not to be considered, i.e. no calculations for this kind of noise have to be done.

The Processing Unit 102 operates as follows: First, the speech signal $d(k)$, which is speech intermingled with background noise, is input to both voice detector 27 and adder 9. If the desired voice (or speech) does not exists, which means $\text{flag2} = 0$, no weight updating happens to the adaptive comb filter 8 and to the adaptive FIR filters 28 and 30. Otherwise, i.e. if speech exists, the weights of all
these adaptive FIR filters are updated by using the least mean square (LMS) algorithm, in which reference and error signals are needed. At the same time the noise detector 26 finds out the existence of different kinds of noise on the basis of inputs, such as $s_i(k)$ and $c_i(k)$ ($i=1,\ldots,M$) from the rotating machine or $r_l(k)$ and $r_r(k)$ from the audio source.

If narrow-band noise does not exist, which means \text{flag0} = 0, the adaptive comb filter 8, which is used to suppress the narrow-band noise would not work, so its output is defined as $y_2(k) = 0$. Otherwise the narrow-band noise is suppressed by adaptive comb filter 8 and adder 9. The output of adder 9 is

$$e_1(k) = d(k) - y_2(k). \quad (1)$$

This output signal $e_1(k)$ is passed on to the next stage. In case that stereo audio signals do not exist, which means \text{flag1} = 0, the adaptive filter 28 does not work, so its output is defined as $y_4(k) = 0$. Otherwise, the reference audio signals $r_l(k)$ and $r_r(k)$ are processed by the adaptive filter 28 and adder 29 so as to suppress the audio signals. The output of adder 29 is

$$e_2(k) = e_1(k) - y_4(k). \quad (2)$$

The output signal $e_2(k)$ is passed on to the last stage of the Processing Unit 102, which comprises the adaptive FIR filter 30 and the adder 31. In this stage, the estimated ambient noise $e_{r2}(k)$ from Ambient Noise Estimator 301 (see Fig.4 for details) is used as reference signal for the adaptive FIR filter 30 so as to suppress the ambient noise. The output of adder 31 is

$$e(k) = e_2(k) - y_5(k). \quad (3)$$

This is the signal with desired speech enhanced and background noise suppressed.
In summary, there are three adaptive filters in this unit to suppress different kinds of background noise. There is narrowband noise, which is dealt with by the adaptive comb filter 8 and adder 9, audio signals from the audio entertainment device, which are coped with by the adaptive filter 28 and adder 29, and ambient noise, which is suppressed by the adaptive FIR filter 30 and adder 31.

**Fig.4** shows the Ambient Noise Estimator 301 in more detail. The function of this Ambient Noise Estimator 301 is to suppress the narrow band noise and the audio signals from the background noise signal $bn(k)$, which consists of narrow band noise, audio signals and ambient noise, so as to provide an approximation of the ambient noise. The Ambient Noise Estimator 301 comprises the adaptive comb filter 6 and the adaptive FIR filter 24. The operation of this Ambient Noise Estimator 301 can be described as follows: At a first stage, if the narrow band noise exists, which means flag0 = 1, the adaptive comb filter 6 and adder 7 are used to suppress the narrow band noise from $bn(k)$. Otherwise the adaptive comb filter 6 would not work, and $y1(k) = 0$. The output of adder 7 is

$$er1(k) = bn(k) - y1(k).$$  \hspace{1cm} (4)

This signal is passed on to a second stage. Here, if the audio signals exist, which means flag1 = 1, the adaptive FIR filter 24 and adder 25 are used to suppress the audio signals from $er1(k)$. In case the audio signals do not exist, which means flag1 = 0, the adaptive FIR filter 24 would not work and $y3(k) = 0$. The output of adder 25 is

$$er2(k) = e1(k) - y3(k).$$  \hspace{1cm} (5)

This signal $er2(k)$ is the approximation of the ambient noise and the output of the Ambient Noise Estimator 301.
In summary, this Ambient Noise Estimator 301 attempts to provide the estimated ambient noise for the noise reduction module by utilizing two adaptive filters 6 and 24. The operation of these adaptive filters 6 and 24 is controlled by the flag signal from the Noise/Audio Signal Detector 26.

The enhanced speech signal from the Processing Unit 102 is output to the D/A converter 32 and sent to the Output Connection Unit 33.

From the foregoing, it can be seen that there has been provided an acoustic noise reduction system/apparatus and a method thereof, particularly useful for suppressing various kinds of noise, so as to improve the speech quality and intelligibility. Incorporated with the communication system, voice activated machinery, broadcast system or monitoring and dispatching system, it is helpful to improve their performance in noisy environment, such as in a car, on a construction site, a factory or an airplane.
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Claims

What is claimed is:

5  1. A system for noise suppression of a speech signal that is intermingled with general noise, said system comprising:
   a) an Input Unit
      (1) for receiving the speech signal intermingled with the general noise (d(k));
      (2) comprising a Sound Processing Unit (4) and a sensor (5) for measuring a rotating rate of a device, in the course of which the Sound Processing Unit (4) generates a first noise signal (s1(k), c1(k)) of the device using the rotating rate value measured by the sensor;
   b) a Processing Unit comprising
      (1) a first adaptive filter (8) to evaluate a dynamically varying second noise signal (y2(k)) out of the first noise signal;
      (2) a first calculation means to evaluate a first noise suppressed signal (e1(k)) out of the second noise signal and the speech signal intermingled with the general noise.

25  2. System as defined in claim 1, wherein said device is a rotating device, particularly an engine.

3. System as defined in claim 1 or 2, wherein said sensor (5) is a means to measure revolutions per minute of the device.

4. System as defined in one of the claims 1 to 3, wherein said first adaptive filter (8) is an adaptive comb filter.
5. System as defined in one of the above claims, wherein said Sound Processing Unit (4) is a sine-wave generator.

6. A System for noise suppression of a speech signal that is intermingled with general noise, said system comprising:
   a) an Input Unit for receiving
      (1) the speech signal intermingled with the general noise (d(k));
      (2) at least one audio signal (rl(k), rr(k)) from an audio source;
   b) a Processing Unit comprising
      (1) a second adaptive filter (28) to evaluate a third noise signal (y4(k)) out of the audio signal;
      (2) a second calculation means to evaluate a second noise suppressed signal (e2(k)) out of the third noise signal and the speech signal intermingled with the general noise.

7. System as defined in claim 6, wherein said Input Unit comprises
   a) a converter (14, 15) to convert said audio signal into a digital audio signal;
   b) a loudspeaker compensation unit (17, 18) that modifies the digital audio signal in such a way that a distortion of the loudspeaker, to which the audio signal is directed, is compensated.

8. System as defined in claim 7, wherein said audio signal is a stereo audio signal and hence for each channel a separate compensation unit and a separate adaptive filter is provided.

9. System as defined in claim 7 or 8, wherein said compensator is constructed to compensate the distortion of the loudspeaker by filtering the digital
audio signal via the transfer function of the loudspeaker.

10. System as defined in claim 9,
wherein said transfer function is evaluated offline.

11. A System for noise suppression of a speech signal that is intermingled with general noise, said system comprising:
   a) an input device as defined in one of the claims 1 to 5 and one of the claims 6 to 10;
   b) a Processing Unit as defined in one of the claims 1 to 5 and one of the claims 6 to 10.
   c) a third calculation means to evaluate a third noise suppressed signal out of the second noise signal (y2(k)), the third noise signal (y4(k)) and the speech signal intermingled with the general noise.

12. System as defined in claim 11,
wherein said Processing Unit comprises
   a) an Ambient Noise Estimator (Fig.4) comprising means to evaluate a forth noise signal (er2(k)) from said audio signal (r1(k), rr(k)), said first noise signal (si(k), ci(k)) and from a background noise signal (bn(k));
   b) a forth calculation means (31) to evaluate a forth noise suppressed signal (e(k)) out of the third noise suppressed signal and the forth noise signal.

13. System as defined in claim 12,
wherein said Processing Unit comprises a third adaptive filter (30) that adjusts the forth noise signal (er2(k)) into a modified forth noise signal (y5(k)).

14. System as defined in one of the claims 12 or 13,
wherein said Ambient Noise Estimator comprises:
   a) a forth adaptive filter (6) that modifies the first noise signal (si(k), ci(k)) into a fifth noise signal (y1(k));
b) a fifth calculation means (7) to evaluate a sixth noise signal \( (er_1(k)) \) from the background noise signal \( (bn(k)) \) and the fifth noise signal \( (y_1(k)) \);

c) a fifth adaptive filter (24) that modifies the audio signal \( (r_1(k), \ r_2(k)) \) into a seventh noise signal \( (y_3(k)) \);

d) a sixth calculation means (25) to evaluate the forth noise signal \( (er_2(k)) \) from the sixth noise signal \( (er_1(k)) \) and the seventh noise signal \( (y_3(k)) \).

15. System as defined in one of the claims 11 to 14,
wherein said Processing Unit comprises a Noise/Audio-Signal-Detector (26) to switch the first adaptive filter (8) if the sensor (5) exceeds a first predefined value and to switch the second adaptive filter (28) if the audio signal exceeds a second predefined value.

16. System as defined in claim 15,
wherein the first predefined value is predefined in such a way that it is exceeded if there is noise from the device.

17. System as defined in claim 15 or 16,
wherein the second predefined value is predefined in such a way that it is exceeded if there is noise from the audio source.

18. System as defined in claim 14 and 15,
wherein said Noise/Audio-Signal-Detector (26) switches the forth adaptive filter (6) and the fifth adaptive filter (24).

19. System as defined in one of the claims 11 to 18,
wherein at least one of the following filters is a FIR-filter: said second adaptive filter (28), said third adaptive filter (30) or said fifth adaptive filter (24).
20. System as defined in one of the claims 13 to 19, wherein said Processing Unit comprises a voice detection unit (27) that switches the first adaptive filter (8), the second adaptive filter (28) and the third adaptive filter (30), if the speech signal intermingled with the general noise (d(k)) exceeds a predefined level.

21. System as defined in one of the above claims, wherein the noise suppressed signal is converted into an analogue signal (32) and transferred to an Output Unit (33).

22. System as defined in one of the above claims, wherein said system is a transceiver.

23. Method for noise suppression in a speech signal intermingled with general noise to be executed on a system as defined in one of the above claims.
Fig.2

1 → 2 → 3 → A/D → d(k)

19 → 20 → 21 → A/D → bn(k)

5 → f0 → Sinewave Generator → s(k) → c(k)

10 → 12 → 14 → A/D → 17 → Left Loudspeaker Compensator → rl(k)

11 → 13 → 15 → A/D → r(k) → Audio Entertainment Device → 16 → 17 → Right Loudspeaker Compensator → rr(k)

FIG. 2
Fig. 3

[Diagram of audio signal processing system with labeled components such as d(k), s(k), c_f(k), r_l(k), r_r(k), b_n(k), Adaptive Comb Filter, Narrowband Noise/Audio Signal Detector, Voice Detector, Ambient Noise Estimator, and Output Connection Unit.]
INTERNATIONAL SEARCH REPORT

A. CLASSIFICATION OF SUBJECT MATTER

IPC 7 G10L21/02

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 7 G10L G10K

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

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)
EPO-Internal, WPI Data, PAJ, INSPEC, COMPENDEX, IBM-TDB

C. DOCUMENTS CONSIDERED TO BE RELEVANT

<table>
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<th>Relevant to claim No.</th>
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| X        | US 5 029 118 A (NAKAJIMA ET AL.)  
column 1 -column 4 | 1-3                  |
| A        | US 5 410 606 A (IMAI ET AL.)  
column 1 -column 3, line 11 | 1                   |
| A        | EP 0 629 054 A (MATSUSHITA ELECTRIC)  
14 December 1994 (1994-12-14)  
page 3, line 37 -page 5, line 5 | 1                   |
| A        | EP 0 654 901 A (TNO)  
page 2 -page 3 | 1                   |

X Further documents are listed in the continuation of box C.

X Patent family members are listed in annex.

* Special categories of cited documents:
*"A" document defining the general state of the art which is not considered to be of particular relevance
*"E" earlier document but published on or after the international filing date
*"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)
*"C" document relating to an oral disclosure, use, exhibition or other means
*"P" document published prior to the international filing date but later than the priority date claimed

Date of the actual completion of the international search 19 July 2001

Date of mailing of the international search report 30.07.2001

Name and mailing address of the ISA European Patent Office, P.B. 5818 Patentlaan 2  
NL - 2280 HT Rijswijk  
Tél. (43270) 340-2030, Tx. 31 655 epo nl,  
Fax: (+31-70) 340-3016

Authorized officer Lange, J
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INTERNATIONAL SEARCH REPORT

Box I Observations where certain claims were found unsearchable (Continuation of item 1 of first sheet)

This International Search Report has not been established in respect of certain claims under Article 17(2)(a) for the following reasons:

1. □ Claims Nos.: because they relate to subject matter not required to be searched by this Authority, namely:

2. □ Claims Nos.: because they relate to parts of the international Application that do not comply with the prescribed requirements to such an extent that no meaningful International Search can be carried out, specifically:

3. □ Claims Nos.: because they are dependent claims and are not drafted in accordance with the second and third sentences of Rule 6.4(a).

Box II Observations where unity of invention is lacking (Continuation of item 2 of first sheet)

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

see additional sheet

1. [X] As all required additional search fees were timely paid by the applicant, this International Search Report covers all searchable claims.

2. □ As all searchable claims could be searched without effort justifying an additional fee, this Authority did not invite payment of any additional fee.

3. □ As only some of the required additional search fees were timely paid by the applicant, this International Search Report covers only those claims for which fees were paid, specifically claims Nos.:

4. □ No required additional search fees were timely paid by the applicant. Consequently, this International Search Report is restricted to the invention first mentioned in the claims; it is covered by claims Nos.:

Remark on Protest □ The additional search fees were accompanied by the applicant's protest.

[X] No protest accompanied the payment of additional search fees.
This International Searching Authority found multiple (groups of) inventions in this international application, as follows:

1. Claims: 4, 21-23 (as far as depending on claim 4):
   Using adaptive comb filters in a system for suppressing periodic noise

2. Claims: 5, 21-23 (as far as depending on claim 5):
   Using a sine-wave generator in a system for suppressing periodic noise

3. Claims: 6-20, 21-23 (as far as depending on claim 6):
   System for suppressing noise from an audio source
### INTERNATIONAL SEARCH REPORT

**Information on patent family members**

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