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(54) **REDUCING NOISE IN AUDIO SYSTEMS**

RAUSCHVERMINDERUNG IN AUDIOSYSTEMEN

REDUCTION DE BRUIT DANS DES SYSTEMES AUDIO

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- **PATENT ABSTRACTS OF JAPAN vol. 2000, no. 22, 9 March 2001 (2001-03-09) -& JP 2001 124621 A (MATSUSHITA ELECTRIC IND CO LTD), 11 May 2001 (2001-05-11)**
- **PATENT ABSTRACTS OF JAPAN vol. 1995, no. 01, 28 February 1995 (1995-02-28) -& JP 06 303689 A (OKI ELECTRIC IND CO LTD), 28 October 1994 (1994-10-28)**

Description

BACKGROUND OF THE INVENTION

Field of the Invention

[0001] The present invention relates to acoustics, and, in particular, to techniques for reducing noise, such as wind noise, generated by turbulent airflow over microphones.

Description of the Related Art

[0002] For many years, wind-noise sensitivity of microphones has been a major problem for outdoor recordings. A related problem is the susceptibility of microphones to the speech jet, i. e., the flow of air from the talker's mouth. Recording studios typically rely on special windscreen socks that either cover the microphone or are placed between the mouth and the microphone. For outdoor recording situations where wind noise is an issue, microphones are typically shielded by acoustically transparent foam or thick fuzzy materials. The purpose of these windscreens is to reduce-or even eliminate--the airflow over the active microphone element to reduceor even eliminate--noise associated with that airflow that would otherwise appear in the audio signal generated by the microphone, while allowing the desired acoustic signal to pass without significant modification to the microphone.

[0003] In patent document US 5,602,963 there is disclosed a speech processing arrangement having at least two microphones. Signals from the microphones are delayed, weighted by weight factors, and summed, where the resulting signal is adaptively filtered to reduce noise components in the microphone signals. WO 95/16259 A discloses a noise reduction system that generates sums and differences of speech signals from different microphones to generate filter coefficients for a Wiener filter used to reduce noise in a combined speech signal. Patent abstracts of Japan vol. 2000, no. 22,9 March 2001 (2001-03-09) -& JP 2001 124621 A (Matsushita Electric Ind Co. Ltd), 11 May 2001 (2001-05-11) disclose a noise eliminating device that applies a fast Fourier transform to a main acoustic signal to predict noise components that are subtracted from the corresponding acoustic frequency spectrum to provide a noise elimination acoustic frequency spectrum.

[0004] JP 06 269084 (D4) discloses a technique for controlling a filter used to reduce noise in audio signals generated by a microphone. In particular, in the context of Fig. 16, D4 teaches a technique for controlling the cut-off frequency of high-pass filter (HPF) 16 to reduce wind noise in the audio signal generated by microphone 11, where controller 33 sets the cut-off frequency of HPF16 based on the output of level ratio sensing circuit 32 (see abstract). Level ratio sensing circuit 32 senses the ratio between the level of audio signal from high-pass filter 31 and the level of the wind noise signal from subtraction circuit 15, where controller 33 sets the cut-off frequency for HPF16 based on the sensed ratio (see, especially, paragraph [0042]).

SUMMARY OF THE INTENTION

[0005] The present invention as defined in claims 1, 2 is related to signal processing techniques that attenuate noise, such as turbulent wind-noise, in audio signals without necessarily relying on the mechanical windscreens of the prior art. In particular, according to certain embodiments of the present invention, two or more microphones generate audio signals that are used to determine the portion of pickup signal that is due to wind-induced noise. These embodiments exploit the notion that wind-noise signals are caused by convective airflow whose speed of propagation is much less than that of the desired acoustic signals. As a result, the difference in the output powers of summed and subtracted signals of closely spaced microphones can be used to estimate the ratio of turbulent convective wind-noise propagation relative to acoustic propagation. Since convective turbulence coherence diminishes quickly with distance, subtracted signals between microphones are of similar power to summed signals. However, signals propagating at acoustic speeds will result in relatively large difference in the summed and subtracted signal powers. This property is utilized to drive a time-varying suppression filter that is tailored to reduce signals that have

[0006] much lower propagation speeds and/or a rapid loss in signal coherence as a function of distance, e.g., noise resulting from relatively slow airflow.

[0007] According to one embodiment, the present invention is a method and an audio system for processing audio signals generated by two or more microphones receiving acoustic signals. A signal processor determines a portion of the audio signals resulting from one or more of (i) incoherence between the audio signals and (ii) one or more audio-signal sources having propagation speeds different from the acoustic signals. A filter filters at least one of the audio signals to reduce the determined portion.

[0008] According to another embodiment, the present invention is a consumer device comprising (a) two or more microphones configured to receive acoustic signals and to generate audio signals; (b) a signal processor configured to determine a portion of the audio signals resulting from one or more of (i) incoherence between the audio signals and (ii) one or more audio-signal sources having propagation speeds different from the acoustic signals; and (c) a filter configured to filter at least one of the audio signals to reduce the determined portion.

[0009] According to yet another embodiment, the present invention is a method and an audio system for processing audio signals generated in response to a sound field by at least two microphones of an audio system. A filter filters the audio signals to compensate for a phase difference between the at least two microphones. A signal processor (1) generates a revised phase difference between the at least two microphones based on the audio signals and (2) updates, based on the revised phase difference, at least one calibration parameter used by the filter.

[0010] In yet another embodiment, the present invention is a consumer device comprising (a) at least two microphones; (b) a filter configured to filter audio signals generated in response to a sound field by the at least two microphones to compensate for a phase difference between the at least two microphones; and (c) a signal processor configured to (1) generate a revised phase difference between the at least two microphones based on the audio signals; and (2) update, based on the revised phase difference, at least one calibration parameter used by the filter.

BRIEF DESCRIPTION OF THE DRAWINGS

[0011] Other aspects, features, and advantages of the present invention will become more fully apparent from the following detailed description, the appended claims, and the accompanying drawings in which like reference numerals identify similar or identical elements.

Fig. 1 shows a diagram of a first-order microphone composed of two zero-order microphones;
Fig. 2 shows a graph of Corcos model coherence as a function of frequency for 2-cm microphone spacing and a convective speed of 5 m/s;
Fig. 3 shows a graph of the difference-to-sum power ratios for acoustic and turbulent signals as a function of frequency for 2-cm microphone spacing and a convective speed of 5 m/s;
Fig. 4 illustrates noise suppression using a single-channel Wiener filter;
Fig. 5 illustrates a single-input/single-output noise suppression system that is essentially equivalent to a system having an array with two closely spaced omnidirectional microphones;
Fig. 6 shows the amount of noise suppression that is applied by the system of Fig. 5 as a function of coherence between the two microphone signals;
Fig. 7 shows a graph of the output signal for a single microphone before and after processing to reject turbulence using propagating acoustic gain settings;
Fig. 8 shows a graph of the spatial coherence function for a diffuse propagating acoustic field for 2-cm spaced microphones, shown compared with the Corcos model coherence of Fig. 2 and for a single planewave;
Fig. 9 shows a block diagram of an audio system, according to one embodiment of the present invention;
Fig. 10 shows a block diagram of turbulent wind-noise attenuation processing using two closely spaced, pressure (omnidirectional) microphones, according to one implementation of the audio system of Fig. 9;
Fig. 11 shows a block diagram of turbulent wind-noise attenuation processing using a directional microphone and a pressure (omnidirectional) microphone, according to an alternative implementation of the audio system of Fig. 9;
Fig. 12 shows a block diagram of an audio system having two omnidirectional microphones, according to an alternative embodiment of the present invention; and
Fig. 13 shows a flowchart of the processing of the audio system of Fig. 12, according to one embodiment of the present invention.

DETAILED DESCRIPTION

Differential Microphone Arrays

[0012] A differential microphone array is a configuration of two or more audio transducers or sensors (e.g., microphones) whose audio output signals are combined to provide one or more array output signals. As used in this specification, the term "first-order" applies to any microphone array whose sensitivity is proportional to the first spatial derivative of the acoustic pressure field. The term "n^{th}-order" is used for microphone arrays that have a response that is proportional to a linear combination of the spatial derivatives up to and including n. Typically, differential microphone arrays combine the outputs of closely spaced transducers in an alternating sign fashion.

[0013] Although realizable differential arrays only approximate the true acoustic pressure differentials, the equations for the general-order spatial differentials provide significant insight into the operation of these systems. To begin, the case for an acoustic planewave propagating with wavevector k is examined. The acoustic pressure field for the planewave case can be written according to Equation (1) as follows:

$$p(\mathbf{k},\mathbf{r},t) = P_o e^{j(\omega t - \mathbf{k}\cdot\mathbf{r})} \tag{1}$$

where P_o is the planewave amplitude, k is the acoustic wavevector, r is the position vector relative to the selected origin, and ω is the angular frequency of the planewave. Dropping the time dependence and taking the n^{th}-order spatial derivative yields Equation (2) as follows:

$$\frac{d^n}{dr^n} p(\mathbf{k},\mathbf{r}) = P_o(-jk\cos\theta)^n e^{-j\mathbf{k}\cdot\mathbf{r}} \tag{2}$$

where θ is the angle between the wavevector k and the position vector r, $r = \|\mathbf{r}\|$, and $k = \|\mathbf{k}\| = 2\pi/\lambda$, where λ is the acoustic wavelength. The planewave solution is valid for the response to sources that are "far" from the microphone array, where "far" means distances that are many times the square of the relevant source dimension divided by the acoustic wavelength. The frequency response of a differential microphone is a high-pass system with a slope of 6n dB per octave. In general, to realize an array that is sensitive to the n^{th} derivative of the incident acoustic pressure field, m n^{th}-order transducers are required, where, $m+p-1=n$. For example, a first-order differential-microphone requires two zero-order sensors (e.g., two pressure-sensing microphones).

[0014] For a planewave with amplitude P_0 and wavenumber k incident on a two-element differential array, as shown in Fig. 1, the output can be written according to Equation (3) as follows:

$$T_1(k,\theta) = P_o\left(1 - e^{-jkd\cos\theta}\right) \tag{3}$$

where d is the inter-element spacing and the subscript indicates a first-order differential array. If it is now assumed that the spacing d is much smaller than the acoustic wavelength, Equation (3) can be rewritten as Equation (4) as follows:

$$\left|T_1(k,\theta)\right| \approx P_o kd\cos\theta \tag{4}$$

[0015] The case where a delay is introduced between these two zero-order sensors is now examined. For a planewave incident on this new array, the output can be written according to Equation (5) as follows:

$$T_1(\omega,\theta) = P_o\left(1 - e^{-j\omega(\tau + d\cos\theta/c)}\right) \tag{5}$$

where τ is equal to the delay applied to the signal from one sensor, and the substitution $k=\omega/c$ has been made, where c is the speed of sound. If a small spacing is again assumed (kd □ π and $\omega\tau$ □ π), then Equation (5) can be written as Equation (6) as follows:

$$\left|T_1(\omega,\theta)\right| \approx P_o\omega\left(\tau + d/c\cos\theta\right) \tag{6}$$

One thing to notice about Equation (6) is that the first-order array has first-order high-pass frequency dependence. The term in the parentheses in Equation (6) contains the array directional response.

[0016] Since n^{th}-order differential transducers have responses that are proportional to the n^{th} power of the wavenumber, these transducers are very sensitive to high wavenumber acoustic propagation. One acoustic field that has high-wavenumber acoustic propagation is in turbulent fluid flow where the convective velocity is much less than the speed of sound. As a result, prior-art differential microphones have typically required careful shielding to minimize the hypersensitivity to wind turbulence.

Turbulent Wind-Noise Models

[0017] The subject of modeling turbulent fluid flow has been an active area of research for many decades. Most of the research has been in underwater acoustics for military applications. With the rapid growth of commercial airline carriers, there has been a great amount of work related to turbulent flow excitation of aircraft fuselage components. Due to the complexity of the equations of motion describing turbulent fluid flow, only rough approximations and relatively simple statistical models have been suggested to describe this complex chaotic fluid flow. One model that describes the coherence of the pressure fluctuations in a turbulent boundary layer along the plane of flow is described in G.M. Corcos, The structure of the turbulent pressure field in boundary layer flows, J. Fluid Mech., 18: pp 353-378, 1964. Although this model was developed for turbulent pressure fluctuation over a rigid half-plane, the simple Corcos model can be used to express the amount of spatial filtering of the turbulent jet from a talker. Thus, this model is used to predict the spatial coherence of the pressure-fluctuation turbulence for both speech jets as well as free-space turbulence.

[0018] The spatial characteristics of the pressure fluctuations can be expressed by the space-frequency cross-spectrum function *G* according to Equation (7) as follows:

$$G_{p_1p_2}(\psi,\omega)=\int_{-\infty}^{\infty}R_{p_1p_2}(\psi,\tau)e^{-j\omega\tau}d\tau \tag{7}$$

where *R* is the spatial cross-correlation function between the two microphone signals, ω is the angular frequency, and ψ is the general displacement variable which is directly related to the distance between measurement points. The coherence function γ is defined as the normalized cross-spectrum by the auto power-spectrum of the two channels according to Equation (8) as follows:

$$\gamma(r,\omega)=\frac{\left|G_{p_1p_2}\right|}{\left[G_{p_1p_1}(\omega)G_{p_2p_2}(\omega)\right]^{1/2}} \tag{8}$$

It is known that large-scale components of the acoustic pressure field lose coherence slowly during the convection with free-stream velocity U, while the small-scale components lose coherence in distances proportional to their wavelengths. Corcos assumed that the stream-wise coherence decays spatially as a function of the similarity variable $\omega r/U_c$, where U_c is the convective speed and is typically related to the free-stream velocity U as $U_c = 0.8U$. The Corcos model can be mathematically stated by Equation (9) as follows:

$$\gamma(r,\omega)=\exp\left(\frac{-\alpha\omega r}{U_c}\right) \tag{9}$$

where α is an experimentally determined decay constant (e.g., α=0.125), and *r* is the displacement (distance) variable. A plot of this function is shown in Fig. 2. The rapid decay of spatial coherence results in the difference in powers between the sums and differences of closely-spaced pressure (zero-order) microphones to be much smaller than for an acoustic planewave propagating along the microphone array axis. As a result, it is possible to detect whether the acoustic signals transduced by the microphones are turbulent-like or propagating acoustic signals by comparing the sum and difference signal powers. Fig. 3 shows the difference-to-sum power ratios (i.e., the ratio of the difference signal power to the sum signal power) for acoustic and turbulent signals for a pair of omnidirectional microphones spaced at 2 cm in a convective fluid flow propagating at 5 m/s. It is clearly seen in this figure that there is a relatively wide difference between the desired acoustic and turbulent difference-to-sum power ratios. The ratio difference becomes more pronounced at low frequencies since the differential microphone output for desired acoustic signals rolls off at -6dB/octave, while the predicted, undesired

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