

AN ACTIVE NOISE CONTROL SYSTEM

CROSS-REFERENCE TO RELATED APPLICATION

[0001] This application claims the benefit of priority to Singapore patent application no. 10202300315Y which was filed on 8 February 2023, the contents of which are hereby incorporated by reference in its entirety for all purposes.

TECHNICAL FIELD

[0002] This application relates to a method of active noise control and an active noise control system that computes a variable penalty factor in real time to constrain an output of the active noise control system to prevent the output of the system from saturating.

BACKGROUND

[0003] There is now increased awareness about health risks associated with urban noise exposure and a greater appreciation of the sound environment in the general public. Excessive levels of noise also cause induced hearing loss, especially for the operators of noisy equipment. In the area of acoustics, various methods are employed to attenuate noise within an auditory space. Noise is most effectively controlled at the source, but control measures are often impeded by the operational requirements around the offending noise source, e.g., the operation of heavy machinery and/or heat management systems. Those skilled in the art have also proposed that the noise be dampened as the noise is propagating. However, these noise suppression techniques are unable to sufficiently reduce the noise to an acceptable level. Due to the physical nature of acoustic waves, passive noise mitigation methods do not efficiently mitigate low-frequency noise, which is the dominant frequency range of most anthropogenic noise sources.

[0004] Therefore, it has been proposed that noise may instead be suppressed through destructive interferences. This can be achieved by overlaying the noise with a compensatory signal that possesses identical amplitude and frequency characteristics but is phase-shifted by 180 degrees, leading to a cancellation or significant reduction of the noise at specific locations within the auditory environment. This destructive interference concept forms the basic principles of existing "active noise cancelling" or "active noise control" (ANC) systems.

[0005] ANC systems typically utilize digital signal processing and digital filtering techniques. These systems will employ a noise sensor, such as a microphone or an equivalent non-acoustic device, to capture an electrical signal corresponding to the unwanted noise. This captured signal is then processed through an adaptive filter to produce an actuator drive signal which in turn is used to drive an acoustic actuator, like a loudspeaker, to create a sound field having a phase opposite to that of the noise. This counteracting sound field effectively diminishes or nullifies the noise in certain areas of the environment. The system may also include a microphone to detect residual noise, generating an "error signal." This error signal is fed back to the adaptive filter, which adjusts its coefficients to minimize the error signal, thereby enhancing noise reduction efficiency.

[0006] Due to the sensitivity of ANC systems to spatiotemporal variability, ANC systems work most effectively when the acoustic space to be controlled is fully characterized, and then integrated into the end-to-end ANC system. Unfortunately, this limits the scalability of an ANC system as it would have been optimized only for a specific scenario and/or environment. The limitation of an ANC system's scalability is further compounded by the necessity for secondary sources to match the high sound pressure levels at low frequencies. Such a situation arises when the ANC system is located adjacent to the noise source. Saturation occurs when the noise to be attenuated exceeds the rated power of the secondary source, which leads to distortion and destabilization of the adaptive algorithm, ultimately also resulting in components of the ANC system failing over time. Most algorithmic solutions to control output saturation either result in severe deterioration of control performance or residual signal quality. Moreover, the computational complexity of the preceding methods severely hampers their real-time practicability in time-critical ANC systems.

[0007] Despite efforts of those skilled in the art to improve the performance of existing ANC systems, existing systems still possess several constraints, such as requiring prior knowledge of the characteristics (e.g., type) of the disturbance/ noise and acoustic path of the disturbance (which varies in real world settings) and having unconstrained output powers which result in distortion and instability of the anti-noise output signal.

SUMMARY

[0008] In one aspect, the present application discloses an active noise control (ANC) system. The ANC system comprises an adaptive filtering module configured to compute estimated disturbance signals, time-dependent filtered reference signals $x'(n)$, and estimated power gains of a secondary path based on a Modified Filtered Least Mean Square (MFxLMS) model when time-dependent reference signals $x(n)$ and an estimated transfer function of the secondary path are provided to the MFxLMS model. The ANC system then generates, for a sample size K of time dependent reference signals $x(n)$, time-dependent variable penalty factors $\alpha(n)$ based on the estimated disturbance signals, the estimated power gains of the secondary path, and a predefined maximum control signal output power, whereby the estimated power gains of the secondary path were generated based on a ratio of the time-dependent filtered reference signals $x'(n)$ and the time-dependent reference signals $x(n)$. The ANC system then proceeds to update a set of filter coefficients $w(n+1)$ of an adaptive control filter of the MFxLMS model by utilizing the generated time-dependent variable penalty factors $\alpha(n)$ to compute a derivative term in a recursive-adaptive model of the adaptive control filter and obtains an output set of control signals $y(n+1)$ by filtering an input set of time-dependent reference signals $x(n+1)$ using the adaptive control filter. A loudspeaker communicatively coupled to the adaptive filtering module is then utilized by the ANC system to generate time-dependent anti-noise audio signals based on the output set of time-dependent control signals $y(n+1)$ as obtained from the adaptive filtering module.

[0009] In another aspect, the present application discloses a method for anti-noise control comprising the steps of computing, using an adaptive filtering module, estimated disturbance signals, time-dependent filtered reference signals $x'(n)$, and estimated power gains of a secondary path based on a Modified Filtered Least Mean Square (MFxLMS) model when time-dependent reference signals $x(n)$ and an estimated transfer function of the secondary path are provided to the MFxLMS model. The method then generates, using the adaptive filtering module, for a sample size K of time dependent reference signals $x(n)$, time-dependent variable penalty factors $\alpha(n)$ based on the estimated disturbance signals, the estimated power gains of the secondary path, and a predefined maximum control signal output power, whereby the estimated power gains of the secondary path were generated based on a ratio of the time-dependent filtered reference signals $x'(n)$ and the time-dependent reference signals $x(n)$. The method subsequently updates, using the adaptive filtering module, a set of filter coefficients

$w(n+1)$ of an adaptive control filter of the MFxLMS model by utilizing the generated time-dependent variable penalty factors $\alpha(n)$ to compute a derivative term in a recursive-adaptive model of the adaptive control filter and obtains, using the adaptive filtering module, an output set of control signals $y(n+1)$ by filtering an input set of time-dependent reference signals $x(n+1)$ using the adaptive control filter. The method then proceeds to generate, using a loudspeaker communicatively coupled to the adaptive filtering module, time-dependent anti-noise audio signals based on the output set of time-dependent control signals $y(n+1)$ as obtained from the adaptive filtering module.

BRIEF DESCRIPTION OF THE DRAWINGS

[0010] Various embodiments of the present disclosure are described below with reference to the following drawings:

Figure 1 illustrates a block diagram of an anti-noise control system in accordance with embodiments of the present disclosure;

Figure 2 illustrates a block diagram representative of a processing system for performing embodiments of the present disclosure;

Figure 3 illustrates a block diagram of a single channel feed forward modified filtered least mean square (MOV-MFxLMS) arrangement in accordance with embodiments of the present disclosure;

Figure 4 illustrates a block diagram of a single channel feedback control system with internal model control (IMC) architecture in accordance with embodiments of the present disclosure;

Figure 5 illustrates a flowchart which sets out the operation of the anti-noise control system in accordance with embodiments of the present disclosure;

Figure 6 illustrates a flowchart of a process for anti-noise control in accordance with embodiments of the present disclosure;

Figure 7(a) illustrates two-weight convergence paths of a conventional filtered least mean square (FxLMS) algorithm;

Figure 7(b) illustrates two-weight convergence paths of a minimum output variance modified filtered least mean square (MOV-MFxLMS) algorithm with an optimal penalty factor;

Figure 7(c) illustrates two-weight convergence paths of a rescaling algorithm;

Figure 7(d) illustrates two-weight convergence paths of a varying penalty factor minimum output variance modified filtered least mean square (MOV-MFxLMS) algorithm in accordance with embodiments of the present disclosure when a sudden varying disturbance is introduced;

Figure 8 illustrates a time history of the variable penalty factor when the power of the disturbance changes at $t = 30s$;

Figure 9 illustrates the disturbance and error signals for various adaptive algorithms; and

Figure 10 illustrates the moving average output power for 1024 samples when active-noise control is implemented using various adaptive algorithms when the maximum output power is set to 0.8.

DETAILED DESCRIPTION

[0011] The following detailed description is made with reference to the accompanying drawings, showing details and embodiments of the present disclosure for the purposes of illustration. Features that are described in the context of an embodiment may correspondingly be applicable to the same or similar features in the other embodiments, even if not explicitly described in these other embodiments. Additions and/or combinations and/or alternatives as described for a feature in the context of an embodiment may correspondingly be applicable to the same or similar feature in the other embodiments.

[0012] In the context of various embodiments, the articles “a”, “an” and “the” as used with regard to a feature or element include a reference to one or more of the features or elements.

[0013] In the context of various embodiments, the term “about” or “approximately” as applied to a numeric value encompasses the exact value and a reasonable variance as generally understood in the relevant technical field, e.g., within 10% of the specified value.

[0014] As used herein, the term “and/or” includes any and all combinations of one or more of the associated listed items.

[0015] As used herein, “comprising” means including, but not limited to, whatever follows the word “comprising”. Thus, use of the term “comprising” indicates that the listed elements are required or mandatory, but that other elements are optional and may or may not be present.

[0016] As used herein, “consisting of” means including, and limited to, whatever follows the phrase “consisting of”. Thus, use of the phrase “consisting of” indicates that the listed elements are required or mandatory, and that no other elements may be present.

[0017] As used herein, “noise” in contrast to a desired audio signal or a useful sound signal, means an unwanted sound or undesirable audio signal generated by offending noise sources, and the noise is not meant to be perceived by a receiver.

[0018] Further, one skilled in the art will recognize that certain functional units in this description have been labelled as modules throughout the specification. The person skilled in the art will also recognize that a module may be implemented as circuits, logic chips or any sort of discrete component. Still further, one skilled in the art will also recognize that a module may be implemented in software which may then be executed by a variety of processor architectures. In embodiments of the disclosure, a module may also comprise computer instructions or executable code that may instruct a computer processor to carry out a sequence of events based on instructions received. The choice of the implementation of the modules is left as a design choice to a person skilled in the art and does not limit the scope of the claimed subject matter in any way.

[0019] A block diagram of an anti-noise control (ANC) system in accordance with embodiments of the present disclosure is illustrated in Figure 1. As illustrated, ANC system 100 comprises controller 104 which is communicatively connected to amplifier and filter module 108, and control unit 102 through either wired or wireless connections means. Controller 104 may comprise a computing module or any other similar computing device. An adaptive filtering module 106 is provided within controller 104 and this module 106 is configured to perform anti-noise digital signal processing methods and/or algorithms, such as, but are not limited to, least mean squares (LMS) methods, modified filtered-x LMS (MFxLMS) methods (where x denotes the input reference signal), or minimum output variance modified-FxLMS (MOV-MFxLMS) methods in accordance with embodiments of the present disclosure.

[0020] Control unit 102 comprises a computing module that is configured to receive inputs from a user of ANC system 100 whereby the user is able to provide input parameters 122 and commands 124 (to turn the ANC system on or off) to controller 104 through the use of this control unit. For example, a user of ANC system 100 may enter a maximum output power limit of ANC system 100 into control unit 102 and this maximum output power limit 122 will then be provided to controller 104 to set an upper limit of ANC system 100. Amplifier and filter module 108 is configured to receive control signals from controller 104 whereby the control

signals are amplified and/or filtered before these control signals are provided to loudspeaker 110. In embodiments of the disclosure, amplifier and filter module 108 may be incorporated into loudspeaker 110. In such an embodiment, the control signals from controller 104 may be provided directly to loudspeaker 110. Upon receiving the control signals from module 108, loudspeaker 110 is configured to generate anti-noise signals to neutralize the noise generated by noise source 115.

[0021] As shown in Figure 1, system 100 also comprises two microphones, microphones 112 and 113. It should be noted that the positions of the microphones are not drawn to scale in Figure 1 and one skilled in the art will recognize that the exact positions of these microphones may be adjusted as required. Microphone 113 is configured to continuously capture reference input signals from noise source 115 and to provide these captured reference input signals to controller 104. As for microphone 112, this microphone is provided adjacent to loudspeaker 110 as microphone 112 may be configured as an “error” sensor for capturing the noise signal from noise source 115 and the anti-noise signals generated by loudspeaker 110.

[0022] A transfer function that represents an acoustic transmission path from loudspeaker 110 to microphone 113 is used when anti-noise digital signal processing methods and/or algorithms are implemented by adaptive filtering module 106. From here on in this description, this acoustic transmission path from loudspeaker 110 to microphone 112 may be known as a “secondary path” of the ANC system.

[0023] In accordance with embodiments of the present disclosure, a block diagram representative of components of processing system 200 that may be provided within controller 104, control unit 102 and/or adaptive filtering module 106 to carry out the digital signal processing functions in accordance with embodiments of the disclosure, or any other modules of the system is illustrated in Figure 2. One skilled in the art will recognize that the exact configuration of each processing system provided within these modules may be different and the exact configuration of processing system 200 may vary and the arrangement illustrated in Figure 2 is provided by way of example only.

[0024] In embodiments of the disclosure, processing system 200 may comprise controller 201 and user interface 202. User interface 202 is arranged to enable manual interactions

between a user and the computing module as required and for this purpose includes the input/output components required for the user to enter instructions to provide updates to each of these modules. A person skilled in the art will recognize that components of user interface 202 may vary from embodiment to embodiment but will typically include one or more of display 240, keyboard 235 and optical device 236.

[0025] Controller 201 is in data communication with user interface 202 via bus 215 and includes memory 220, processing unit or processor 205 mounted on a circuit board that processes instructions and data for performing the method of this embodiment, an operating system 206, an input/output (I/O) interface 230 for communicating with user interface 202 and a communications interface, in this embodiment in the form of a network card 250. Network card 250 may, for example, be utilized to send data from these modules via a wired or wireless network to other processing devices or to receive data via the wired or wireless network. Wireless networks that may be utilized by network card 250 include, but are not limited to, Wireless-Fidelity (Wi-Fi), Bluetooth, Near Field Communication (NFC), cellular networks, satellite networks, telecommunication networks, Wide Area Networks (WAN) and etc.

[0026] Memory 220 and operating system 206 are in data communication with processor 205 via bus 210. The memory components include both volatile and non-volatile memory and more than one of each type of memory, including Random Access Memory (RAM) 223, Read Only Memory (ROM) 225 and a mass storage device 245, the last comprising one or more solid-state drives (SSDs). One skilled in the art will recognize that the memory components described above comprise non-transitory computer-readable media and shall be taken to comprise all computer-readable media except for a transitory, propagating signal. Typically, the instructions are stored as program code in the memory components but can also be hardwired. Memory 220 may include a kernel and/or programming modules such as a software application that may be stored in either volatile or non-volatile memory.

[0027] Herein the term “processor” is used to refer generically to any device or component that can process such instructions and may include: a microprocessor, a processing unit, a microcontroller, a programmable logic device or other computational device. That is, processor 205 may be provided by any suitable logic circuitry for receiving inputs, processing them in accordance with instructions stored in memory and generating outputs (for example to the

memory components or on display 240). In this embodiment, processor 205 may be a single core or multi-core processor with memory addressable space. In one example, processor 205 may be multi-core, comprising—for example—an 8 core CPU. In another example, it could be a cluster of CPU cores operating in parallel to accelerate computations.

[0028] Figure 3 illustrates a single channel adaptive feedforward ANC model based on a modified-FxLMS arrangement 300 that may be implemented by adaptive filtering module 106. As shown in Figure 3, adaptive control filter 302 which may be defined by $\omega(n) = [\omega_0(n) \ \omega_1(n) \ \dots \ \omega_{I-1}(n)]^T$ contains I finite-impulse-response (FIR) coefficients. When a time-dependent reference signal $x(n)$ is provided to arrangement 300, adaptive control filter 302 will process the time-dependent reference signal $x(n)$ to generate a control signal $y(n)$. In embodiments of the disclosure, the time-dependent reference signal $x(n)$ may be obtained from microphone 113 as illustrated in Figure 1.

[0029] The control signal $y(n)$ then propagates through secondary path s , forming anti-noise signal $y'(n)$ that reaches error sensor 304. In embodiments of the disclosure, the control signal $y(n)$ may be used by loudspeaker 110 (as shown in Figure 1) to generate the anti-noise signals and error sensor 304 may be configured to receive audio signals from microphone 112 (as illustrated in Figure 1). Hence, it can be said that the anti-noise signal $y'(n)$ comprises anti-noise signals from loudspeaker 110 that have been modified according to the transfer function of the secondary path s , i.e., the acoustic transmission path from loudspeaker 110 to microphone 112.

[0030] Estimated secondary path $\hat{s}$ comprises a digital model of the secondary path s and is used to predict how the anti-noise signal will be altered as it travels through the secondary path. In other words, the estimated secondary path $\hat{s}$ may be used with the control signal $y(n)$ to produce an estimated anti-noise signal $\hat{y}'(n)$.

[0031] In embodiments of the disclosure, the error signal $e(n)$ comprises the difference between the anti-noise signal $y'(n)$ and a disturbance signal $d(n)$, where the disturbance signal $d(n)$ may be generated by noise source 115. The cost function J of arrangement 300 may then be formulated based on the minimum output variance to constrain the output power of arrangement 300 and the cost function J is defined as:

$$J = \mathbb{E}[e_m^2(n)] + \alpha \mathbb{E}[y^2(n)] \quad \dots(1)$$

where $\mathbb{E}[\cdot]$ and α ($\alpha \geq 0$) represent an expectation operator and a penalty factor respectively, and $e_m(n)$ is defined as a modified error signal. The modified error signal $e_m(n)$ is defined as:

$$e_m(n) = \hat{d}(n) - \hat{y}'_m(n) \quad \dots(2)$$

where $\hat{d}(n)$ is defined as the estimated disturbance signal and $\hat{y}'_m(n)$ is defined as the modified anti-noise signal. The estimated disturbance signal $\hat{d}(n)$ may then be defined as:

$$\hat{d}(n) = e(n) + \sum_{j=0}^{J-1} \hat{s}_j \omega^T(n-j) x(n-j) \quad \dots(3)$$

while the modified anti-noise signal $\hat{y}'_m(n)$ may be defined as:

$$\hat{y}'_m(n) = \sum_{i=0}^{I-1} \sum_{j=0}^{J-1} \omega_i(n) \hat{s}_j x(n-i-j) \quad \dots(4)$$

[0032] In embodiments of the disclosure, the secondary path s may be defined as $s = [s_0 \ s_1 \ \dots \ s_{J-1}]^T$ and the estimated secondary path $\hat{s}$ may be defined as $\hat{s} = [\hat{s}_0 \ \hat{s}_1 \ \dots \ \hat{s}_{J-1}]^T$.

[0033] Based on Figure 3, it can be seen that the time dependent reference signal $x(n)$ that has been processed by the transfer function of the secondary path $\hat{s}$ may be defined as time dependent filtered reference signal $x'(n)$. To recap, adaptive control filter 302 will process the time-dependent reference signal $x(n)$ to generate a control signal $y(n)$. The control signal $y(n)$ and the time dependent filtered reference signal $x'(n)$ may then be respectively defined as:

$$\begin{aligned} y(n) &= \omega^T(n) x(n) \\ x'(n) &= \sum_{j=0}^{J-1} \hat{s}_j x(n-j) \end{aligned} \quad \dots(5)$$

[0034] When the derivative of the cost function J in Equation (1) is set to '0', an optimal control filter ω_{opt} (of control filter 302) may be derived as:

$$\omega_{opt} = (R_{x'x'} + \alpha R_{xx})^{-1} r_{x'\hat{d}} \quad \dots(6)$$

where $R_{x'x'}$ and R_{xx} are defined as the auto-correlation matrices and $r_{x'\hat{d}}$ is defined as the correlation vector. The auto-correlation matrices $R_{x'x'}$ and R_{xx} , and the correlation vector $r_{x'\hat{d}}$ may then be defined respectively as:

$$\begin{aligned} R_{xx} &= \mathbb{E}[x(n)x^T(n)] \\ R_{x'x'} &= \mathbb{E}[x'(n)x'^T(n)] \\ r_{x'\hat{d}} &= \mathbb{E}[x'(n)\hat{d}(n)] \end{aligned} \quad \dots(7)$$

[0035] Based on a stochastic gradient descent method to minimize the instantaneous value of the cost function J as defined in Equation (1), a recursive formula for the adaptive control filter 302 may then be defined as:

$$\omega(n+1) = \omega(n) + \mu[x'(n)e_m(n) - \alpha x(n)y(n)] \quad \dots(8)$$

[0036] In embodiments of the disclosure, Equation (8) may be defined as the minimum output variance modified-FxLMS (MOV-MFxLMS) algorithm where μ is defined as the constant adaptation step size of the algorithm and $[x'(n)e_m(n) - \alpha x(n)y(n)]$ is defined as a derivative term of Equation (8). Equation (8) may be adaptively computed by LMS module 308 to obtain the coefficients of adaptive control filter 312 and once computed, LMS module 308 may then update adaptive control filter 302 accordingly.

[0037] Based on Equation (6), it can be seen that the penalty factor α has to be carefully selected as the penalty factor α directly affects the attenuation performance of arrangement 300. When the penalty factor α is too large, this causes the minimization of the term $\mathbb{E}[y^2(n)]$ in Equation (1) to be overemphasized and this results in a reduction of the gain of control filter 302 and as a result, decreases the attenuation performance of arrangement 300. Conversely, when the penalty factor α is too small, it will not be able to effectively constrain the control signal $y(n)$, and this causes the output power of the system to saturate.

[0038] The optimal penalty factor that satisfies a power output constraint can be derived by solving the equation relating to the optimal control filter, i.e., Equation (6), using a Lagrangian optimization problem approach and this optimization problem may be formulated as:

$$\min_{\omega} \xi(\omega) = \mathbb{E}[e_m^2(n)] \quad \dots(9a)$$

$$s.t. \quad g(\omega) = \mathbb{E}[y^2(n)] - \rho^2 \leq 0 \quad \dots(9b)$$

where ρ^2 is defined as the maximum control signal output power allowed.

[0039] Hence, the Lagrangian function may then be formulated as:

$$\mathcal{L}(\omega, \lambda) = \mathbb{E}[e_m^2(n)] + \lambda \{\mathbb{E}[y^2(n)] - \rho^2\} \quad \dots(10)$$

where λ ($\lambda \geq 0$) is defined as the Lagrange multiplier.

[0040] When the derivative of the Lagrangian function is set to zero, i.e., $\Delta\mathcal{L}(\omega, \lambda) = 0$, the optimal control filter may then be derived as:

$$\omega_0 = (R_{x'x'} + \lambda_0 R_{xx})^{-1} r_{x'd} \quad \dots(11)$$

where λ_0 is defined as the optimal Lagrange factor. The optimal Lagrange factor λ_0 yields solutions to the constrained and unconstrained conditions and this may be defined as:

$$\lambda_0 = \frac{\mathbb{E}[\hat{d}(n)\hat{y}'_{mo}(n)] - \mathbb{E}[\hat{y}'^2_{mo}(n)]}{\rho^2} \text{ or } 0 \quad \dots(12)$$

with $\hat{d}(n)$ and $\hat{y}'^2_{mo}(n)$ being used to replace $d(n)$ and $y_0'^2(n)$ respectively.

[0041] It should be noted that the optimal control filter as described in Equation (6) is comparable to the Lagrangian formulation in Equation (11) when $\omega_{opt} = \omega_0$, which in turn relates the optimal penalty factor $\alpha_0 = \lambda_0$. To remove the cross-correlation term $\mathbb{E}[\hat{d}(n)\hat{y}'_{mo}(n)]$ from Equation (12), an additional condition, suitable for use in an ANC

system, was imposed where $\hat{d}(n)$ and $\hat{y}'_{mo}(n)$ were positively correlated:

$$\frac{\mathbb{E}[\hat{d}(n)\hat{y}'_{mo}(n)]}{\sqrt{\mathbb{E}[\hat{d}^2(n)]\mathbb{E}[\hat{y}'_{mo}{}^2(n)]}} = 1 \quad \dots(13)$$

where $\mathbb{E}[\hat{d}(n)] = \mathbb{E}[\hat{y}'_{mo}(n)] = 0$. The optimal penalty factor α_0 is then derived as:

$$\alpha_0 = \lambda_0 = G_s \sqrt{\frac{\sigma_{\hat{d}}^2}{\rho^2 G_s}} - 1 \quad \dots(14)$$

where G_s is defined as the power gain of the secondary path s . The power gain of the secondary path G_s is then defined as:

$$G_s = \frac{\sigma_{\hat{y}'_{mo}}^2}{\rho^2} \quad \dots(15)$$

where $\sigma_{\hat{d}}^2 = \mathbb{E}[\hat{d}^2(n)]$ is defined as the variance of the estimated disturbance and $\sigma_{\hat{y}'_{mo}}^2 = \mathbb{E}[\hat{y}'_{mo}{}^2(n)]$ is defined as the optimally constrained anti-noise.

[0042] As $\hat{d}(n)$ is known, the optimal penalty factor α_0 may be predicted over time and this enables adaptive filtering module 106 to utilize arrangement 300 to attain the desired output power constraint, even if the noise type or amplitude changes throughout the control process.

[0043] In order to obtain a variable penalty factor $\alpha(n)$, a varying power gain $\hat{G}_s(n)$ of the secondary path s has to be firstly obtained. This can be done by utilizing the time-dependent reference signal $x(n)$ and the time-dependent filtered reference signal $x'(n)$, as the reference signal $x(n)$ contains the same frequency components as the disturbance signal in most real-world ANC applications. Based on the above, the varying power gain $\hat{G}_s(n)$ of the secondary path s may be defined as:

$$\hat{G}_s(n) = \frac{\sigma_{x'}^2(n)}{\sigma_x^2(n)} \approx G_s \quad \dots(16)$$

where $\sigma_{x'}^2(n)$ is defined as the variance in the time-dependent filtered reference signal $x'(n)$ and $\sigma_x^2(n)$ is defined as the variance in the time-dependent reference signal $x(n)$.

[0044] When the time-dependent reference signal $x(n)$ and the time-dependent filtered reference signal $x'(n)$ as utilized, this causes the varying power gain $\hat{G}_s(n)$ of the secondary path s to update in real-time, according to the variation in the disturbance or noise signal. Furthermore, as the time-dependent reference signal $x(n)$ and the time-dependent filtered reference signal $x'(n)$ are obtained when the MFxLMS algorithm is performed by arrangement 300, no additional convolution steps need to be carried out in order to compute the varying power gain $\hat{G}_s(n)$ of the secondary path s .

[0045] Additionally, in embodiments of the disclosure, when local stationarity is assumed for the time-dependent reference signal $x(n)$ and estimated disturbance signal $\hat{d}(n)$ within K samples, where K may comprise, but is not limited to, a few thousand samples (e.g., $K=3,200$ samples when an averaging process is performed at 2 second intervals at a sampling rate of 16,000 Hertz), the power gain of the secondary path G_s can be estimated in real-time by the following equation:

$$\hat{G}_s(n) = \frac{\max \left\{ \sum_{k=0}^{K-1} x'^2(n-k), \varepsilon_1 \right\}}{\max \left\{ \sum_{k=0}^{K-1} x^2(n-k), \varepsilon_2 \right\}} \quad \dots(17)$$

where the variable penalty factor $\alpha(n)$ may be defined by:

$$\alpha(n) = \max \left\{ \hat{G}_s(n) \left(\sqrt{\frac{\sum_{k=0}^{K-1} \hat{d}^2(n-k)}{K \rho^2 \hat{G}_s(n)}} - 1 \right), 0 \right\} \quad \dots(18)$$

where ε_1 is defined as a first threshold parameter and ε_2 is defined as a second threshold parameter to ensure that the numerator and denominator values of Equation (18) are non-zero values. In embodiments of the disclosure, the first threshold parameter and the second threshold parameter each comprise a value between $1e^{-6}$ and $1e^{-10}$. In order to prevent the variable penalty factor $\alpha(n)$ from becoming a negative value, a minimum threshold value of '0' is set in Equation (18) for the max() function. The variable penalty factor $\alpha(n)$ is computed in every

iteration of the adaptive algorithm in arrangement 300 as such, this ensures that arrangement 300 is able to satisfy the ANC system's output power constraints even though the disturbance signals may fluctuate and change with each iteration.

[0046] By substituting Equation (18) into the recursive formula for the adaptive control filter at Equation (8) (or the MOV-MFxFMS algorithm), the recursive formula for the adaptive control filter with the variable penalty factor may thus be defined as:

$$\omega(n+1) = \omega(n) + \mu[x'(n)e_m(n) - \alpha(n)x(n)y(n)] \quad \dots(19)$$

[0047] In embodiments of the disclosure, LMS module 308 may then be configured to compute the coefficients of adaptive control filter 302 using Equation (19) and once computed, update adaptive control filter 302 accordingly. This ensures that the control signal $y(n)$ will meet the system's output power constraints without saturating, regardless of fluctuations or changes to the disturbance signal $d(n)$.

[0048] Based on the above, with reference to Figure 1, it is understood that controller 104 will capture the time-dependent reference signal $x(n)$ using microphone 113 for each iteration. This captured time-dependent reference signal $x(n)$ will then be utilized by adaptive filtering module 106 together with an estimated transfer function of the secondary path $\hat{s}$ to compute estimated disturbance signals $\hat{d}(n)$, time-dependent filtered reference signals $x'(n)$, and estimated power gains of a secondary path $\hat{G}_s(n)$ based on the Modified Filtered Least Mean Square (MFxFMS) model shown in Figure 3.

[0049] Once these values have been obtained, adaptive filtering module 106 may then proceed to generate, for K samples of the time-dependent reference signals $x(n)$, time-dependent variable penalty factors $\alpha(n)$. This may be done based on previously computed estimated disturbance signals $\hat{d}(n)$, the estimated power gains of the secondary path $\hat{G}_s(n)$, and a maximum control signal output power ρ^2 as provided by control unit 102. In this embodiment of the disclosure, the estimated power gains of the secondary path $\hat{G}_s(n)$ were generated based on a ratio of the time-dependent filtered reference signals $x'(n)$ and the time-dependent reference signals $x(n)$.

[0050] After adaptive filtering module 106 has computed the time-dependent variable penalty factor $\alpha(n)$, this value is then used to compute a derivative term in a recursive-adaptive model of the adaptive control filter and this derivative term is then used to update a set of filter coefficients $\omega(n+1)$ of the adaptive control filter of the MFxLMS model shown in Figure 3.

[0051] An output set of control signals $y(n+1)$ are then obtained by adaptive filtering module 106 filtering an input set of time-dependent reference signals $x(n+1)$ using the updated adaptive control filter. Loudspeaker 110 then proceeds to generate time-dependent anti-noise audio signals based on the output set of time-dependent control signals $y(n+1)$ as obtained from adaptive filtering module 106. This process then repeats itself until control unit 102 is triggered by the user of system 100 to end the anti-noise cancellation process.

[0052] Figure 4 illustrates another embodiment of the disclosure whereby the minimum output variance (MOV) algorithm is employed in a generic feedback algorithm with an Internal Model Control (IMC) architecture. When such a feedback algorithm is utilized, a reference microphone, such as microphone 113 (as illustrated in Figure 1), is not required to be utilized. Instead, only microphone 112 (as illustrated in Figure 1) is utilized to capture the error signal which is then used to compute estimated disturbance signals $\hat{d}(n)$, which comprises a replacement of the reference signal as originally obtained by microphone 112. The reference signal $\hat{x}(n)$ would then be processed by the FxLMS algorithm, by convoluting the reference signal $\hat{x}(n)$ with the estimated secondary path to produce the time-dependent filtered reference signal $\hat{x}'(n)$. The time-dependent filtered reference signal $\hat{x}'(n)$ is then used to compute the weights of feedback control filter 402 through LMS 408.

[0053] A flowchart which sets out the operation of the anti-noise control system illustrated in Figures 1 and 3 in accordance with embodiments of the present disclosure is illustrated in Figure 5. In embodiments of the disclosure, process 500 as illustrated in Figure 5 may be performed by controller 104 and/or adaptive filtering module 106.

[0054] Process 500 begins at step 502, with process 500 performing a check to determine if a calibration process is to be performed by process 500. If process 500 determines at step 502 that a calibration process is required, process 500 will proceed to step 504.

[0055] At step 504, a secondary path calibration process will be initialized whereby process 500 will cause the loudspeaker of the anti-noise system to generate Gaussian white noise. Process 500 will then acquire the signal from the error sensor microphone and compare this acquired signal with the generated Gaussian white noise using a Normalized Least Mean Squares (NLMS) algorithm at step 506. The objective of the NLMS algorithm is to adaptively adjust the coefficients of the adaptive control filter in the MFxLMS model so that the output of the filter closely matches the captured signal. This iterative step takes place at step 508. In general, this process is based on minimizing the mean square error between the output of the adaptive control filter and the actual received signal. The NLMS algorithm updates the coefficients of the adaptive control filter in a way that normalizes the updates by the power of the input. This normalization provides robustness to variations in the input signal's power, leading to faster convergence and more stable operation in varying acoustic environments. As the NLMS algorithm iteratively adjusts and converges its coefficients, the filter's response becomes an increasingly accurate representation of the secondary path. Once process 500 has determined that the set of filter coefficients have converged, process 500 then proceeds to step 510. At this step, the set of filter coefficients have converged and may then be stored by process 500 as a transfer function of the estimated secondary path model $\hat{s}$ in a database to be retrieved at a later stage as required. Process 500 then proceeds to the ANC initialization stage at step 514.

[0056] Conversely, if process 500 determines at step 502 that the calibration process is not required, process 500 will proceed to retrieve the estimated secondary path model $\hat{s}$ as stored in the database. This takes place at step 512. Process 500 then proceeds to the ANC initialization stage at step 514. During this step, process 500 will set up and configure the system's components, such as microphones and loudspeaker, and also proceed to initialize the adaptive filter.

[0057] At step 516, process 500 determines if the user had turned on the ANC. If the ANC is turned on, process 500 will proceed to step 518. At step 518, process 500 will retrieve the current value of the desired maximum output power ρ^2 set by a user of the system. Process 500 will then recompute, at step 520, the estimated disturbance signals $\hat{d}(n)$ and the variance of the estimated disturbance signals σ_d^2 from the MFxLMS model.

[0058] At step 522, process 500 then computes the estimated power gains of the secondary path $\hat{G}_s(n)$ from the MFxLMS model and the time-dependent variable penalty factor $\alpha(n)$. Process 500 then determines at step 524, if the variance of the estimated disturbance signals σ_d^2 exceeds a predetermined threshold. In embodiments of the disclosure, the value of the predetermined threshold is dependent on the signal-to-noise ratio of the microphones and/or the number of bits in the digital-to-analogue converter used by process 500. If process 500 determines that the variance of the estimated disturbance signals σ_d^2 exceeds the predetermined threshold, process 500 will compute, at step 526, a set of filter coefficients $\omega(n+1)$ of the adaptive control filter of the MFxLMS model and use this adaptive control filter to generate an output set of control signals $y(n+1)$ by filtering an input set of time-dependent reference signals $x(n+1)$. The output set of control signals are then provided to the loudspeaker by process 500 at step 530 to generate the anti-noise signals.

[0059] Returning to step 524, if process 500 determines that the variance of the estimated disturbance signals σ_d^2 does not exceed the predetermined threshold, process 500 will set the value of the control signal to zero '0' and will use the set of filter coefficients $\omega(n)$ of the previous iteration for the adaptive control filter and use this filter to generate the output signal for the loudspeaker.

[0060] Returning further to step 516, if process 500 determines that the ANC is turned off, process 500 will proceed to step 528 where the value of the control signal is set to zero '0' and process 500 will use the set of filter coefficients $\omega(n)$ of the previous iteration for the adaptive control filter. Process 500 then uses this filter to generate the output signal for the loudspeaker.

[0061] After the anti-noise signal has been generated by the loudspeaker, process 500 then returns to step 516 and the steps then repeat themselves as described above.

[0062] A process for anti-noise control is illustrated in Figure 6 whereby process 600 may be carried out by controller 104 and/or adaptive filtering module 106. Process 600 begins at step 602 where estimated disturbance signals, time-dependent filtered reference signals $x'(n)$, and estimated power gains of a secondary path based on a Modified Filtered Least Mean Square (MFxLMS) model are computed when time-dependent reference signals $x(n)$ and an estimated transfer function of the secondary path are provided to the MFxLMS model.

[0063] At step 604, time-dependent variable penalty factors $\alpha(n)$ based on the estimated disturbance signals, the estimated power gains of the secondary path, and a predefined maximum control signal output power are generated by process 600 whereby the estimated power gains of the secondary path were generated based on a ratio of the time-dependent filtered reference signals $x'(n)$ and the time-dependent reference signals $x(n)$. In embodiments of the disclosure, this is done for a sample size K of time-dependent reference signals $x(n)$.

[0064] At step 606, a set of filter coefficients $\omega(n+1)$ of an adaptive control filter of the MFxLMS model are updated by process 600 by utilizing the generated time-dependent variable penalty factors $\alpha(n)$ to compute a derivative term in a recursive-adaptive model of the adaptive control filter.

[0065] Process 600 then obtains an output set of control signals $y(n+1)$ by filtering an input set of time-dependent reference signals $x(n+1)$ using the adaptive control filter. This takes place at step 608.

[0066] At step 610, process 600 then generates, using a loudspeaker communicatively coupled to the adaptive filtering module, time-dependent anti-noise audio signals based on the output set of time-dependent control signals $y(n+1)$ as obtained from the adaptive filtering module. Process 600 then ends.

[0067] In embodiments of the disclosure, in the step of generating the estimated power gain of the secondary path by process 600, process 600 may compute for the sample size K of time dependent reference signals $x(n)$, the estimated power gain of the secondary path based on a computed sum of a square of the time-dependent filtered reference signals $x'(n)$ when it is determined that the computed sum of the square of the time-dependent filtered reference signals $x'(n)$ exceeds a first threshold parameter, and based on a computed sum of a square of the time-dependent reference signals $x(n)$ when it is determined that the computed sum of the square of the time-dependent reference signals $x(n)$ exceeds a second threshold parameter.

[0068] In embodiments of the disclosure, the derivative term of the recursive-adaptive model of the adaptive control filter may be computed by process 600 based on the time-dependent filtered reference signals $x'(n)$, modified error signals, the time-dependent reference signals $x(n)$, and the time-dependent control signals $y(n)$, whereby the time-dependent control signals $y(n)$ and the modified error signals were computed based on the MFxLMS model when the time-dependent reference signals $x(n)$ and the estimated transfer function of the secondary path were provided to the MFxLMS model.

[0069] In embodiments of the disclosure, before process 600 generates the time-dependent anti-noise signals, process 600 computes a variance of the estimated disturbance signals; and filters the input set of time-dependent reference signals $x(n+1)$ using the adaptive control filter that has been updated with a previous set of filter coefficients $\omega(n)$ to obtain the output set of control signals $y(n+1)$ when it is determined that the computed variance of the estimated disturbance signals is below a predetermined threshold.

[0070] In embodiments of the disclosure, in the step of providing the estimated transfer function of the secondary path to the MFxLMS model, process 600 may generate using the loudspeaker, Gaussian white noise and retrieve from a database, a stored transfer function of the secondary path that corresponds with an error signal associated with the generated Gaussian white noise, wherein the stored transfer function is utilized as the estimated transfer function of the secondary path of the MFxLMS model.

[0071] In embodiments of the disclosure, the first threshold parameter and the second threshold parameter each may comprise a value between $1e^{-6}$ and $1e^{-10}$.

[0072] Numerical Simulation

[0073] A numerical simulation comparison was performed to compare the performance of the variable penalty factor MOV-MFxLMS algorithm in accordance with embodiments of the disclosure and other algorithms under various conditions.

[0074] *A. Convergence path due to a sudden change in disturbance*

[0075] In this simulation, the performance of a MOV-MFxLMS algorithm with the proposed variable penalty factor was compared with a MOV-FxLMS algorithm with a predetermined fixed penalty factor, a rescaling algorithm with a maximum output magnitude parameter of $y_{max} = 3.46$ as obtained through trial and error, and the conventional FxLMS algorithm, when a sudden varying disturbance is provided to all these algorithms. In particular, a reference signal containing a bandlimited white noise (800–7200 Hz) will undergo a sudden change in its signal power from $\sigma_x^2 = 0.305$ to $\sigma_x^2 = 0.540$ at the 30s mark. The optimal penalty factors for the two disturbances were computed as $\alpha_1 = 0.0461$ and $\alpha_2 = 0.3255$, respectively. The sampling rate, step-size, primary and secondary path were set as $F_s = 16 \text{ kHz}$, $\mu = 0.0002$, $p = [1.62 \ 0.41]^T$ and $s = [0.03 \ 0.87]^T$. Figure 7 illustrates the convergence path of the control filter as compared between the MOV-MFxLMS algorithm, the MOV-FxLMS algorithm, and the unconstrained FxLMS method.

[0076] In particular, Figure 7(a) illustrates two-weight convergence paths of a conventional filtered least mean square (FxLMS) algorithm when a sudden varying disturbance is introduced. As can be seen in this Figure, the FxLMS algorithm converges to the optimal control filter $\omega_{opt} = [1.62 \ 0.41]^T$ while ignoring the subjected power constraint of $\rho^2 = 1$, giving the optimal power output for the control signal of $\sigma_{y_0}^2 = 1.1214$ for the first 30s of noise and $\sigma_{y_0}^2 = 1.9769$ for the next 30s.

[0077] Figure 7(b) illustrates two-weight convergence paths of a minimum output variance modified filtered least mean square (MOV-MFxLMS) algorithm with an optimal penalty factor when a sudden varying disturbance is introduced. However, it is interesting to note that while only one of the two optimal penalty factors was used throughout the simulation, it managed to converge to the first optimal constrained filter in the first 30s when the corresponding α_1 was used but failed to converge to the optimal constrained filter in the subsequent 30s with a louder disturbance, violating the power output constraint by having $\sigma_{y_0}^2 = 1.7630$. On the other hand, the fixed penalty factor of α_2 yielded a similar observation where convergence occurred only in the corresponding final 30s. Although the constraint was not violated in the first 30s when lower amplitude disturbance was introduced, it led to an over-constrained output of $\sigma_{y_0}^2 = 0.5673$.

[0078] Figure 7(c) illustrates two-weight convergence paths of a rescaling algorithm when a sudden varying disturbance is introduced. While the rescaling algorithm shown in Figure 7(c) showed a change in its convergence path after the 30s mark, it can be seen that both paths do not converge to its respective optimal constrained filter as the maximum amplitude of the control signal was constrained, which is difficult to translate to its equivalent maximum output power for non-tonal noise.

[0079] Figure 7(d) illustrates two-weight convergence paths of a varying penalty factor minimum output variance modified filtered least mean square (MOV-MFxLMS) algorithm in accordance with embodiments of the present disclosure when a sudden varying disturbance is introduced. The MOV-MFxLMS algorithm with the variable penalty factor and $K = 256$ is able to mitigate the problem faced by the rescaling algorithm as shown in Figure 7(c). As illustrated in Figure 7(d), it can be seen that it initially converges to the first optimal constrained filter (path marked by circles) and shifts to the second optimal constrained filter (path marked by triangles) when the second noise is present, thereby satisfying the power-output requirement for the duration of the simulation.

[0080] The same phenomena can be observed in Figure 8, which illustrates a time history of the variable penalty factor when the power of the disturbance changes at $t = 30$ s. Due to the moving-average operation, the variable penalty factor fluctuates around α_1 at the first 30s and around α_2 at the next 30s. Therefore, this simulation demonstrates that the proposed variable penalty factor MOV-MFxLMS algorithm is able to achieve optimal constraint control even if the disturbance has a sudden change in amplitude. The same conclusion can be extended to the variation of the primary noise signal.

[0081] *B. Active control of real noise based on measured paths.*

[0082] To further verify its effectiveness, the proposed varying MOV-MFxLMS algorithm was utilized to attenuate a compounding noise on the primary and secondary path measured from an air duct. Figures 9 and 10 depict the error signal and its moving average output power when four distinct noise stages are provided. The first 30s contain a 400–600 Hz broadband noise, and a larger amplitude of 400–600 Hz broadband noise is compounded to the existing

noise for the next 30s. Similarly, a measured construction noise, having a dominant 70–400 Hz noise, and aircraft flyover noise, having a dominant 400–3500 Hz noise, is compounded to its previous noise for the 60–90 s and 90–120 s time frame respectively. The output saturation effect in the control signal is not considered in this simulation when the output power constraint is exceeded. The FxLMS algorithm has the most noise reduction as shown in Figure 9, which plots the amplitude of the error signal over time, while Figure 10 illustrates the moving average output power for 1024 samples when active-noise control is implemented using various adaptive algorithms when the maximum output power is set to 0.8, i.e., the moving average output power when its output power constraint was ignored, especially in the fourth stage where the output power increases tremendously to an approximate peak value of $\sigma_y^2(n) \approx 10$ due to the non-stationary aircraft noise. For MOV-FxLMS with a fixed penalty factor calibrated for the 30–60 s noise, it can be seen that it only managed to meet the power output constraint during the 30–60 s period but has exceeded the output constraint during the 60–120 s period due to the additional construction noise and aircraft noise.

[0083] In contrast, the output power for the 0–30s period has been over-constrained as it showed a lower output power than that of the FxLMS algorithm. This issue can be mitigated by the varying penalty factor MOV-MFxLMS algorithm that managed to maintain the power output constraint for the 30–120 s period as shown in Figure 10. It was found to have the same convergence behavior as FxLMS for the first 30s as the estimated moving averaged disturbance power is less than $K\rho^2\hat{G}_s(n)$ ($K = 1024$), leading to the calculated penalty factor from to be ‘0’. This verifies the moving average estimation method which satisfies the output power constraint due to a sudden change in noise type.

[0084] Numerous other changes, substitutions, variations, and modifications may be ascertained by the skilled in the art and it is intended that the present application encompass all such changes, substitutions, variations, and modifications as falling within the scope of the appended claims.

CLAIMS

1. An active noise control (ANC) system comprising:
 - an adaptive filtering module configured to:
 - compute estimated disturbance signals, time-dependent filtered reference signals $x'(n)$, and estimated power gains of a secondary path based on a Modified Filtered Least Mean Square (MFxLMS) model when time-dependent reference signals $x(n)$ and an estimated transfer function of the secondary path are provided to the MFxLMS model;
 - generate, for a sample size K of time dependent reference signals $x(n)$, time-dependent variable penalty factors $\alpha(n)$ based on the estimated disturbance signals, the estimated power gains of the secondary path, and a predefined maximum control signal output power, whereby the estimated power gains of the secondary path were generated based on a ratio of the time-dependent filtered reference signals $x'(n)$ and the time-dependent reference signals $x(n)$;
 - update a set of filter coefficients $\omega(n+1)$ of an adaptive control filter of the MFxLMS model by utilizing the generated time-dependent variable penalty factors $\alpha(n)$ to compute a derivative term in a recursive-adaptive model of the adaptive control filter; and
 - obtain an output set of control signals $y(n+1)$ by filtering an input set of time-dependent reference signals $x(n+1)$ using the adaptive control filter; and
 - a loudspeaker communicatively coupled to the adaptive filtering module, the loudspeaker being configured to generate time-dependent anti-noise audio signals based on the output set of time-dependent control signals $y(n+1)$ as obtained from the adaptive filtering module.

2. The active noise control system according to claim 1, wherein the generation of the estimated power gain of the secondary path by the adaptive filtering module comprises the adaptive filtering module being configured to:
 - compute, for the sample size K of time dependent reference signals $x(n)$, the estimated power gain of the secondary path based on a computed sum of a square of the time-dependent filtered reference signals $x'(n)$ when it is determined that the computed sum of the square of the time-dependent filtered reference signals $x'(n)$ exceeds a first threshold parameter, and based on a computed sum of a square of the time-dependent reference signals $x(n)$ when it is determined that the computed sum of the square of the time-dependent reference signals $x(n)$ exceeds a second threshold parameter.

3. The active noise control system according to any one of claims 1 or 2, wherein the derivative term of the recursive-adaptive model of the adaptive control filter is computed based on the time-dependent filtered reference signals $x'(n)$, modified error signals, the time-dependent reference signals $x(n)$, and the time-dependent control signals $y(n)$, whereby the time-dependent control signals $y(n)$ and the modified error signals were computed based on the MFxLMS model when the time-dependent reference signals $x(n)$ and the estimated transfer function $\hat{S}$ of the secondary path were provided to the MFxLMS model.

4. The active noise control system according to any one of claims 1 to 3, whereby before the loudspeaker is configured to generate the time-dependent anti-noise signals, the adaptive filtering module is further configured to:

compute a variance of the estimated disturbance signals; and

filter the input set of time-dependent reference signals $x(n+1)$ using the adaptive control filter that has been updated with a previous set of filter coefficients $\omega(n)$ to obtain the output set of control signals $y(n+1)$ when it is determined that the computed variance of the estimated disturbance signals is below a predetermined threshold.

5. The active noise control system according to any one of claims 1 to 4, whereby the provision of the estimated transfer function of the secondary path to the MFxLMS model further comprises the adaptive filtering module being configured to:

cause the loudspeaker to generate Gaussian white noise; and

retrieve, from a database, a stored transfer function of the secondary path that corresponds with an error signal associated with the generated Gaussian white noise, wherein the stored transfer function is utilized as the estimated transfer function of the secondary path of the MFxLMS model.

6. The active noise control system according to claim 2 wherein the first threshold parameter and the second threshold parameter each comprise a value between $1e^{-6}$ and $1e^{-10}$.

7. A method for active noise control comprising:

computing, using an adaptive filtering module, estimated disturbance signals, time-dependent filtered reference signals $x'(n)$, and estimated power gains of a secondary path

based on a Modified Filtered Least Mean Square (MFxLMS) model when time-dependent reference signals $x(n)$ and an estimated transfer function of the secondary path are provided to the MFxLMS model;

generating, using the adaptive filtering module, for a sample size K of time dependent reference signals $x(n)$, time-dependent variable penalty factors $\alpha(n)$ based on the estimated disturbance signals, the estimated power gains of the secondary path, and a predefined maximum control signal output power, whereby the estimated power gains of the secondary path were generated based on a ratio of the time-dependent filtered reference signals $x'(n)$ and the time-dependent reference signals $x(n)$;

updating, using the adaptive filtering module, a set of filter coefficients $\omega(n+1)$ of an adaptive control filter of the MFxLMS model by utilizing the generated time-dependent variable penalty factors $\alpha(n)$ to compute a derivative term in a recursive-adaptive model of the adaptive control filter; and

obtaining, using the adaptive filtering module, an output set of control signals $y(n+1)$ by filtering an input set of time-dependent reference signals $x(n+1)$ using the adaptive control filter; and

generating, using a loudspeaker communicatively coupled to the adaptive filtering module, time-dependent anti-noise audio signals based on the output set of time-dependent control signals $y(n+1)$ as obtained from the adaptive filtering module.

8. The method according to claim 7, wherein the step of generating the estimated power gain of the secondary path comprises the steps of:

computing, using the adaptive filtering module, for the sample size K of time dependent reference signals $x(n)$, the estimated power gain of the secondary path based on a computed sum of a square of the time-dependent filtered reference signals $x'(n)$ when it is determined that the computed sum of the square of the time-dependent filtered reference signals $x'(n)$ exceeds a first threshold parameter, and based on a computed sum of a square of the time-dependent reference signals $x(n)$ when it is determined that the computed sum of the square of the time-dependent reference signals $x(n)$ exceeds a second threshold parameter.

9. The method according to any one of claims 7 or 8, wherein the derivative term of the recursive-adaptive model of the adaptive control filter is computed based on the time-dependent filtered reference signals $x'(n)$, modified error signals, the time-dependent

reference signals $x(n)$, and the time-dependent control signals $y(n)$, whereby the time-dependent control signals $y(n)$ and the modified error signals were computed based on the MFxLMS model when the time-dependent reference signals $x(n)$ and the estimated transfer function of the secondary path were provided to the MFxLMS model.

10. The method according to any one of claims 7 to 9, whereby before step of generating the time-dependent anti-noise signals, the method further comprises the steps of:

computing, using the adaptive filtering module, a variance of the estimated disturbance signals; and

filtering, using the adaptive filtering module, the input set of time-dependent reference signals $x(n + 1)$ using the adaptive control filter that has been updated with a previous set of filter coefficients $\omega(n)$ to obtain the output set of control signals $y(n + 1)$ when it is determined that the computed variance of the estimated disturbance signals is below a predetermined threshold.

11. The method according to any one of claims 7 to 10, whereby the step of providing the estimated transfer function of the secondary path to the MFxLMS model further comprises the steps of:

generating, using the loudspeaker, Gaussian white noise; and

retrieving, using the adaptive filtering module, from a database, a stored transfer function of the secondary path that corresponds with an error signal associated with the generated Gaussian white noise, wherein the stored transfer function is utilized as the estimated transfer function of the secondary path of the MFxLMS model.

12. The method according to claim 8 wherein the first threshold parameter and the second threshold parameter each comprise a value between $1e^{-6}$ and $1e^{-10}$.