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Noise Cancellation In Headphones

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Abstract

The active noise reducing headphone is probably the most successful application of active control of sound – the technology of canceling sound with sound. This report presents an outlined technical review of noise cancellation in headphones. The principles of passive noise attenuation are presented after which active attenuation is introduced showing how the two complement the attenuation performance. Both the analog and digital implementations of the noise cancellation system are described, including a briefing on the possible combination of both into a single system.

Introduction

Active noise control (ANC) is a method for reducing the unwanted disturbances by the introduction of controllable secondary sources, whose outputs are arranged to interfere destructively with the disturbance from the original primary source. In order to obtain good cancellation, it is generally important that the secondary source is adjusted to compensate for changes in the primary noise source.

Noise cancellation in headphones relies on the acoustic isolation characteristic of headphones with active noise reduction. By their nature, headphones block out some degree of external noise because the ear-cups absorb it, but active noise control goes a step further and diminish the noise that manages to get through. Active headphones are used mainly in highly noisy environments to protect the user from the excessive noise. Such headphones usually use both passive and active noise attenuation. Passive attenuation takes place when the incoming sound is blocked or attenuated by the headphone shell covering the ear. This is most effective at high frequencies. In the active attenuation of sound a loudspeaker placed inside the headphone shell produces the anti-noise signal thereby actively canceling the external noise. This works well at low frequencies. "A good headphone will effectively combine low frequency active attenuation with high frequency passive attenuation to provide high attenuation of the external noise at a wide frequency range." [Raf02]

Passive attenuation

One instance of passive attenuation is when the headset shell is sealed to the head using an appropriate cushion therefore blocking the sound. Since the cushion needs to be soft or flexible to allow for a good fit and a tight seal, it also allows the shell to vibrate when exposed to external sound. The vibrations of the shell then radiate sound into the shell cavity, which is then perceived by the listener. Bulky headphones with stiff cushions are more difficult to vibrate and therefore provide better passive attenuation.

A more better passive attenuation can be achieved if the headphones are designed as shown in Fig. 2, where the electronics are the bandpass filters and audio limiters that reduce noise by stripping the high and low frequencies in the audio signal and flattening the amplitudes before being output to the headphone transducer or loudspeaker. However the drastic bandpass filtering in these devices does not clean up noise embedded in the passband and because the microphone is located outside the earcup, the sampled noise is not a perfect replica of the noise inside the earcup, which is altered by passing through the earcup as well as by internal reflections. Therefore, in some situations, the anti-noise signal may actually introduce noise inside the headphones.

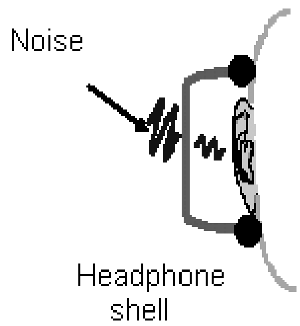

Fig. 1 Passive headphone attenuation “from [Raf02]”

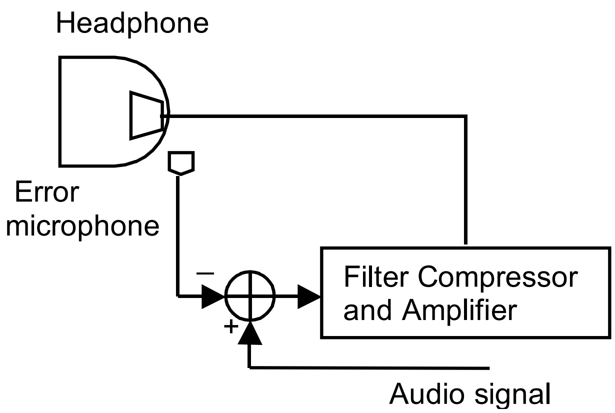

Fig 2 - Noise filtering headphone “adapted from [Moy99]”

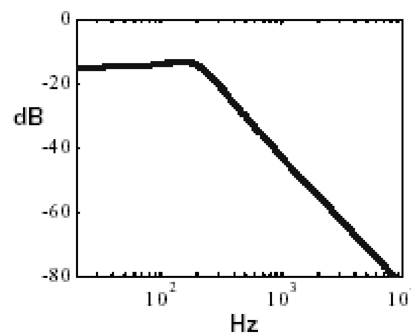

Fig .3 Typical passive attenuation “from [Raf02]

The passive attenuation of a typical headphone is illustrated in Fig. 3, showing about 15 dB of attenuation below the cutoff frequency, and an increasing attenuation above the cutoff frequency. At high frequencies the amount of attenuation is dependent upon the quality of the seal from the ear cushion around the ear, the construction of the ear cups, the sound absorbent materials used in the ear cups and the electronic elements used for filtering and audio limiting. "In practice imperfect seal will degrade the attenuation at low frequencies, while high frequency dynamics of the shell and its cavity will affect the attenuation at higher frequencies." [Raf02]

Active attenuation

Analog feedback control

A more accurate anti-noise signal is possible if the microphone is placed inside the earcup in front of the loudspeaker as shown in Fig. 4. In this case the circuit is electrically a closed one because the microphone now samples both the sound emitted by the loudspeaker as well as the noise that enter the headphone shell. This signal is fed back to the desired audio signal input as an error correction for the noise. The system is thus characterized by negative feedback, which is highly popular in control systems.

Applications that utilize a feedback structure often use analog devices for the implementation of the active headphone system. Traditional controller design methods use modular filter elements, such as lead, lag and notch filters, appropriately tuned, to

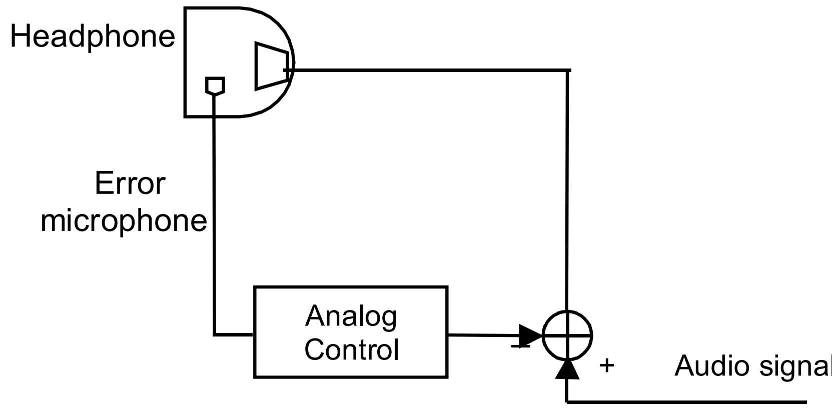

Fig 4. Active headphone attenuation with feedback control " from [Moy99]"

shape the response of the system to attain good performance with sufficient stability margins. Headphones, which employ such methods, are prone to oscillation due to possibility of excessive phase shift of the feedback signal caused by the time delay from the distance between the microphone and the loudspeaker and the inherent delays of the microphone and loudspeaker. Recently, various design methods have been developed which attempt to provide a better trade-off between performance and stability by taking into account in more detail the plant uncertainty. Some of these methods such as H-infinity, Internal Model Control and Quantitative Feedback Theory have also been used in active headset applications [Raf02]. The attenuation performance of an analog active headset is defined in the design stage, so a fixed attenuation curve will be achieved irrespective of the spectrum of the incoming noise. This limits the applicability of the analog active headphones to work in different noisy environments.

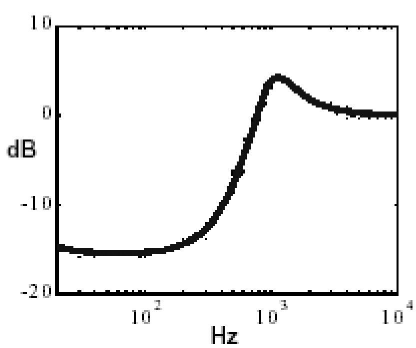

Fig. 5 Typical active attenuation "from [Raf02] "

Digital feedforward control

Modern active noise canceling headphones generally use feedforward structure, which employ digital filters to generate control signals. "Digital filters used for noise cancellation are called adaptive filters and can correct for both phase and amplitude errors." [Moy99]

In the feed-forward ANC headphones as shown in Fig. 6, any reference noise that enters the headphone independent of the audio signal, is picked up by a noise reference microphone and adaptively filtered and fed to a loudspeaker mounted in the headphone set. So there are two headphones in a feedforward structure. The loudspeaker emits an anti-noise signal, also referred to as the secondary noise, to cancel the primary noise in the region inside the headphone shell. The residual noise is picked up by an error microphone, which is mounted in the headphone set, closed to the loudspeaker, and this error signal is used to tune the filter driving the loudspeaker. The tuning of the filter is done in the absence of the audio signal, and in the presence of the audio signal, the filter is tuned to its last updated coefficients.

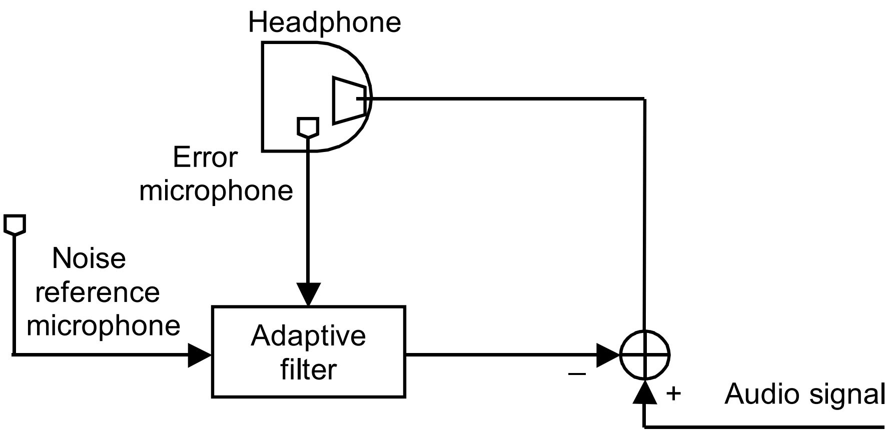

Fig 6 Active headphones with feedforward control “adapted from [Wan97]

Active feedforward control of sound rely on the timely detection of the reference noise signal which needs to be filtered and transmitted to the loudspeaker as an anti-noise signal in time to cancel the propagating primary noise. Excessive delay in the digital control path means that the cancellation signal will arrive too late to perform cancellation. The delay in the digital controller is generally the overall delay of the system including the sampling delay in the DSP and DACs, and the phase delay of the low-pass filters. If the total electric delay exceeds the acoustic delay from the reference microphone to the loudspeaker, then the optimal filter will be non-causal, and prediction will be required to attenuate broadband signals. In this case only bandlimited or predictable signals can be successfully attenuated. Performance is thus limited to narrow band or tonal noise when using conventional DSP systems.

Adaptive algorithms used in ANC headphones

ANC headphones employ the adaptive digital filter to generate control signals. This is partly due to the computational complexities involved in designing the optimal filter and also due to time-varying properties of the acoustic paths. The adaptive filter updates its coefficients iteratively to track the best possible solutions. The well-known algorithms used in noise canceling headphones are detailed in this section.

LMS Algorithm

In many practical applications the statistics about the reference signal received at the reference microphone from the primary source and the signal at the error sensor are unknown. Also there are more than one disturbance sources and so there should be multiple-channel feedforward systems that control stochastic or random disturbances. Time domain formulation is needed in such cases.

Active control at a number of error sensors is achieved by detecting the waveform of the primary sources with a number of noise sensors, and feeding these signals through a matrix of control filters to a set of secondary sources. Fig 7 shows the typical block diagram of such a case. "It is assumed that there are K noise sensors, and thus K reference signals, M secondary sources which are loudspeakers for headphones and L

error sensors where $L \geq M$ is assumed (only single element for each of the sources and signals shown in Fig 7, assumed repetitive). The K reference signals are fed to a matrix of adaptive filters whose outputs are used to drive M secondary sources, with output signals $y_m(n)$. The (m, k)-th filter, which is assumed time invariant for the time being, has coefficients, w_{mki} so that the output from the m-th secondary source can be expressed as" [Ell93]

$$y_m(n) = \sum_{k=1}^{K} \sum_{i=0}^{I-1} w_{mki} x_k(n-i) \tag{1}$$

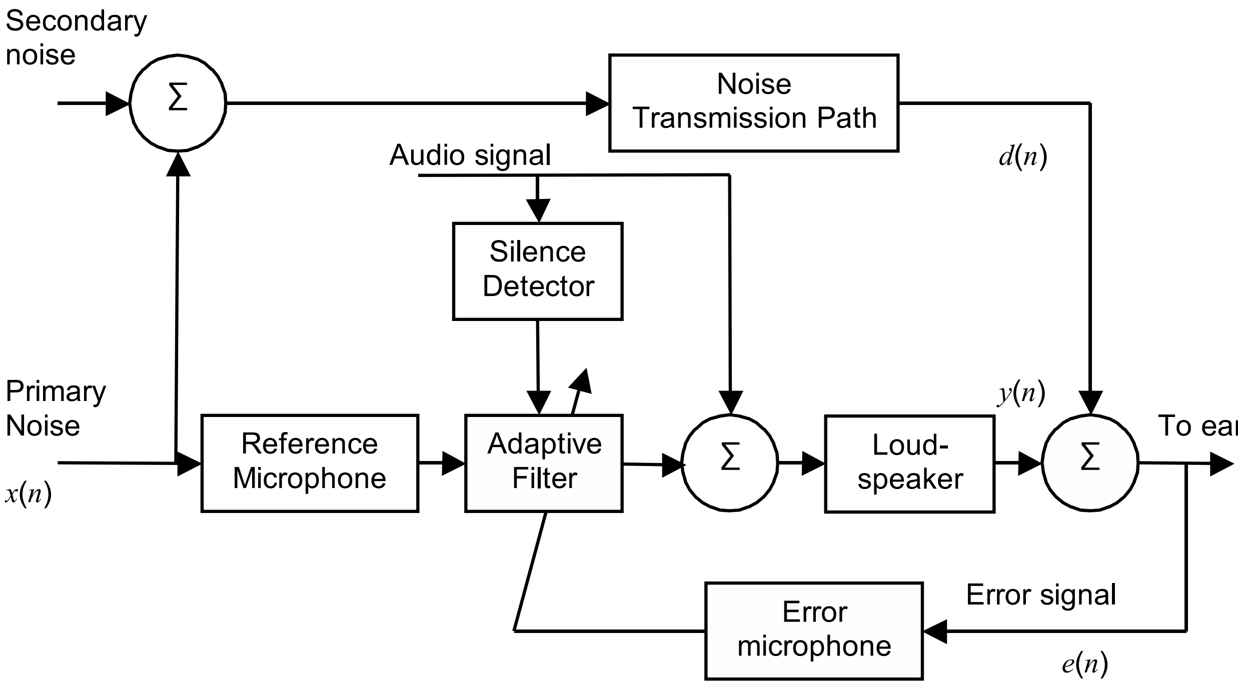

Fig. 7 Block diagram of noise cancellation using LMS algorithm "adapted from [Wan97]"

"Each control filter is linearly coupled to each of L error sensors, with outputs $e_l(n)$, via secondary paths which can be modeled as (fixed) J-th order FIR filters (where J can be as large as necessary), so that

$$e_l(n) = d_l(n) + \sum_{m=1}^{M} \sum_{j=0}^{J-1} c_{lmj} y_m(n-j) \tag{2}$$

where c_{lmj} are the coefficients of the (l,m)–th filter and $d_l(n)$ is the secondary noise at the l–th error sensor in the absence of control" [Ell93], i.e. due to the primary field, not detected by the reference microphone. Substituting equation (1) into (2) gives

$$e_l(n) = d_l(n) + \sum_{m=1}^{M} \sum_{i=0}^{I-1} \sum_{k=1}^{K} \sum_{j=0}^{J-1} c_{lmj} w_{mki} x_k(n-i-j) \tag{3}$$

which may be rewritten as a single summation over the number of control filter coefficients (MKI) as

$$e_l(n) = d_l(n) + \sum_{m=1}^{M}\sum_{i=0}^{I-1}\sum_{k=1}^{K} w_{mki} r_{lmk}(n-i-j) \qquad (4)$$

where $r_{lmk}(n)$ is the k-th reference signal filtered by the response of the path from the m-th secondary source to the l-th error sensor.

$$r_{lmk}(n) = \sum_{j=0}^{J-1} c_{lmj} x_k(n-i) \qquad (5)$$

We now seek the stochastic gradient algorithm, which adjusts all the control filter coefficients to minimize the instantaneous cost function equal to the sum of the squared signals at the error sensors:

$$J(n) = \sum_{l=1}^{L} e_l^2(n) \qquad (6)$$

The derivative of $J(n)$ with respect to the general control filter coefficient w_{mki} is

$$\frac{\partial J(n)}{\partial w_{mki}} = 2\sum_{l=1}^{L} e_l(n)\frac{\partial e_l(n)}{\partial w_{mki}} = 2\sum_{l=1}^{L} e_l(n) r_{lmk}(n-i) \qquad (7)$$

where the final expression follows from equation (4). Updating each filter coefficient by an amount proportional to $-\frac{\partial J(n)}{\partial w_{mki}}$ at every sample time leads to a simple form of LMS algorithm.

$$w_{mki}(n+1) = w_{mki}(n) - \alpha \sum_{l=1}^{L} e_l(n) r_{lmk}(n-i) \qquad (8)$$

where α is a convergence coefficient.

"Since multiple channels are assumed, the equation (8) is called a Multiple Error LMS algorithm. The success of the control algorithm depends on a number of factors including whether

i) the reference signals persistently excite the control filters so that ill-conditioning is avoided,
ii) the FIR model of each secondary path can be accurately measured so that the true filtered reference signals can be generated,
iii) the speed of the adaptation of the control filter coefficients is sufficiently slow so as not to invalidate the assumption that the control filters are time invariant." [Ell93]

The main drawback of the LMS algorithm is the speed of convergence that gets very slow if there is a big spread among the eigenvalues of **R**(a small eigenvalue leads to a small correction in the filter weights so that there should be a lot of iterations to reach the optimal point in that dimension). The another disadvantage of LMS filters is that they must be retrained for changes in the feedback path (eg. Temperature changes, a different person wears the headphones).

Filtered-x LMS algorithm

In the ideal situation the error signal is found as the difference between the primary noise signal previously referred to as $d(n)$ and the estimated signal $y(n)$ in the absence of the audio signal. In the case of ANC in headsets this is not the case. As mentioned in the last sub-section, using adaptive filtering for ANC in headsets involves a number of physical transducers. One of these is the loudspeaker acting as a secondary source, which can cause problems, as the transfer function of this must be inverted. The most obvious solution to the problem is to place the inverse filter in series with the loudspeaker and hereby cancel the effect of it. However, loudspeakers are in general not minimum phase systems and are therefore not invertible. The problem can be solved by filtering the input signal to the adaptive control algorithm by an estimate of the transfer function of the loudspeaker. The adaptive algorithm can both be the LMS algorithm or the normalized LMS algorithm, which is a variant of LMS algorithm. This modification results in the filtered-x LMS algorithm.

In this section, an adaptive feedback model using filtered-x LMS algorithm is discussed. The basic idea of the adaptive feedback ANC system is to estimate the primary noise and use it as a reference signal for the adaptive filter since the detection headphone in this feedback scheme unlike in feedforward scheme is not available.

The error signal is expressed as

$$\begin{aligned} e(n) &= d(n) - \hat{y}(n) \\ &= d(n) - s(n) * y(n) \\ &= d(n) - s(n) * \left[\mathbf{w}^{\mathbf{T}}\mathbf{(n)x(n)}\right] \end{aligned} \tag{9}$$

where $s(n)$ is the impulse response of the secondary path at time n,

$$\mathbf{w}(n) = \begin{bmatrix} w_0 & w_1 & \ldots & w_{L-1} \end{bmatrix}^T \tag{10}$$

is the coefficient vector of W(z) at time n,

$$\mathbf{x}(\mathrm{n}) = [x(n) \quad x(n\text{-}1) \quad \ldots \quad x(n\text{-}L+1)]^{\mathrm{T}} \tag{11}$$

is the signal vector at time n and L is the order of the filter W(z).

The objective of the adaptive filter is to minimize the instantaneous squared error,

$$\hat{\xi}(n) = e^2(n)$$

The most widely used method to achieve this is the stochastic gradient or LMS algorithm, which updates the coefficient in the negative gradient direction with step size μ [Kuo96]:

$$w(n+1) = w(n) - \frac{\mu}{2}\nabla\hat{\xi}(n) \tag{12}$$

where $\nabla\hat{\xi}(n)$ is an instantaneous estimate of the MSE gradient at time n, and can be expressed as

$$\nabla\hat{\xi}(n) = \nabla^2 e(n) = 2[\nabla e(n)]e(n) \tag{13}$$

From equation (9)

$$\nabla e(n) = -s(n) * \mathbf{x}(n) = -\hat{\mathbf{x}}(n) \tag{14}$$

where $\hat{\mathbf{x}}(n) = \begin{bmatrix} \hat{x}(n) & \hat{x}(n-1) & \ldots & \hat{x}(n-L+1) \end{bmatrix}^T$ (15)

Therefore the gradient estimate becomes

$$\nabla\hat{\xi}(n) = -2\hat{\mathbf{x}}(n)e(n) \tag{16}$$

Substituting equation (16) into equation (12),

$$w(n+1) = w(n) + \mu\hat{\mathbf{x}}(n)e(n) \tag{17}$$

As seen from the equation the filtered version of the $x(n)$ is used to update the controller weight vector, hence the name Filtered-x LMS algorithm. In practical ANC applications, $S(z)$ is unknown and must be estimated by an additional filter, $\hat{S}(z)$ Therefore the filtered

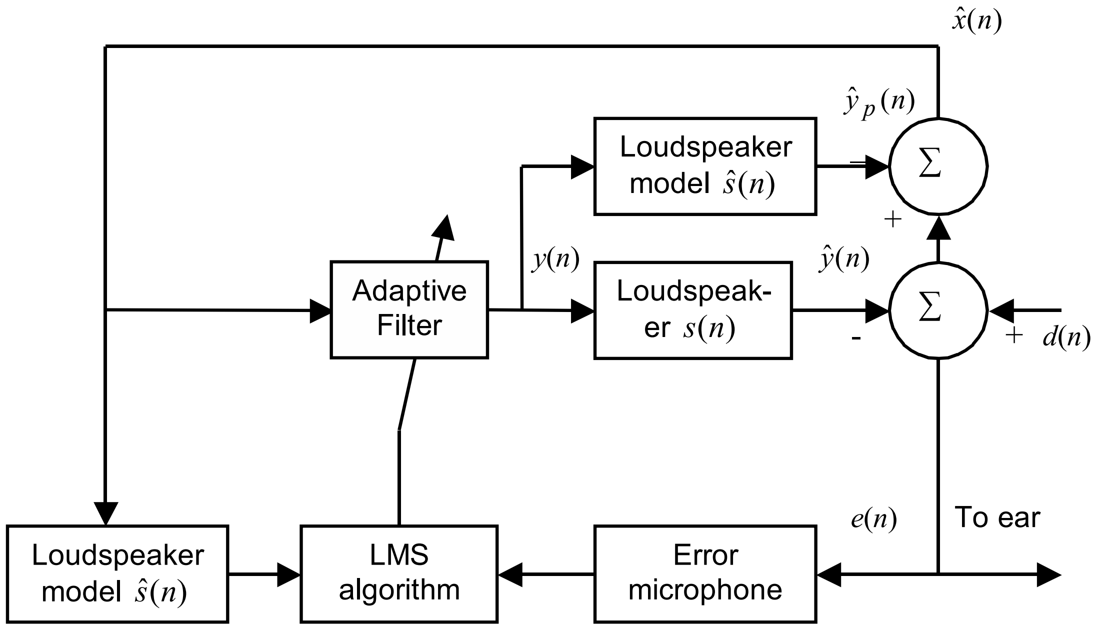

Fig. 8 - Block diagram of the ANC feedback configuration of the filtered-x LMS algorithm "adapted from [Wan97]"

reference is generated by passing the reference signal through this estimate of the secondary path. The algorithm now remains the same as in equation except with

$$\hat{\mathbf{x}}(n) = \hat{s}(n) * \mathbf{x}(n) \tag{18}$$

where $-\hat{\mathbf{x}}(n)$ is the estimated impulse response of the secondary path filter, $\hat{S}(z)$ [Kuo96]. The filtered-x LMS algorithm converges when the step-size is sufficiently small, and the loudspeaker model is sufficiently close to the true loudspeaker.

The implementation of the filtered-x LMS algorithm is somewhat more complicated than that of the normal LMS algorithm because of the need to generate the filtered reference signal. Also the stability of the algorithm depends on the accuracy of the estimated filter $\hat{S}(z)$ modeling the true secondary path.

In the report by Wang and Tse [Wan97], the filtered-x LMS algorithm was implemented on a T1 TMS320C30 DSP that drives a Sony CD550 headphone/microphone system. The experiment indicates that adaptive control results in a dramatic improvement in performance over fixed control for narrow band noise [Wan97].

Adjoint LMS algorithm
The adjoint algorithm was introduced as a low computational alternative to filtered-x LMS algorithm [Wan96]. In adjoint LMS algorithm, the error signal $e(n)$ rather than the reference noise signal $x(n)$ is filtered through an adjoint (reverse) version of the loudspeaker model. The adjoint algorithm is given by

$$\begin{aligned} w(n+1) &= w(n) + \mu \mathbf{x}(n-M+1)\hat{e}(n-M+1) \\ \hat{e}(n) &= s(n) * e(n) \end{aligned} \qquad (19)$$

where M is the order of the filter. In the paper by Wan [Wan96], the experiment for noise cancellation for a general case using adjoint LMS algorithm shows a greater range of stability versus the learning curve. Also it was observed that adjoint LMS algorithm has an equivalent rate of convergence and misadjustment to filtered-x LMS algorithm with a substantial computational savings over Multiple Error LMS algorithm [Wan96].

Combined analog and digital system

To achieve noise cancellation over a wide range of frequencies, the analog feedback control and digital feedforward control can be combined in one system, broadband performance achieved by analog system, and tracking of time varying narrowband noise by the digital system. This combined method has been demonstrated for headphones by Winberg and Carne in 1999 [Raf02].

The analog feedback controller C is now combined with the adaptive feedforward controller W by adding their control outputs at the loudspeaker input as shown in Fig. 9. One way to view the new system is by considering the plant together with the analog controller as the new plant, controlled by the digital system [Raf02]. The analog controller therefore controls the plant P, while the digital controller controls the whole system.

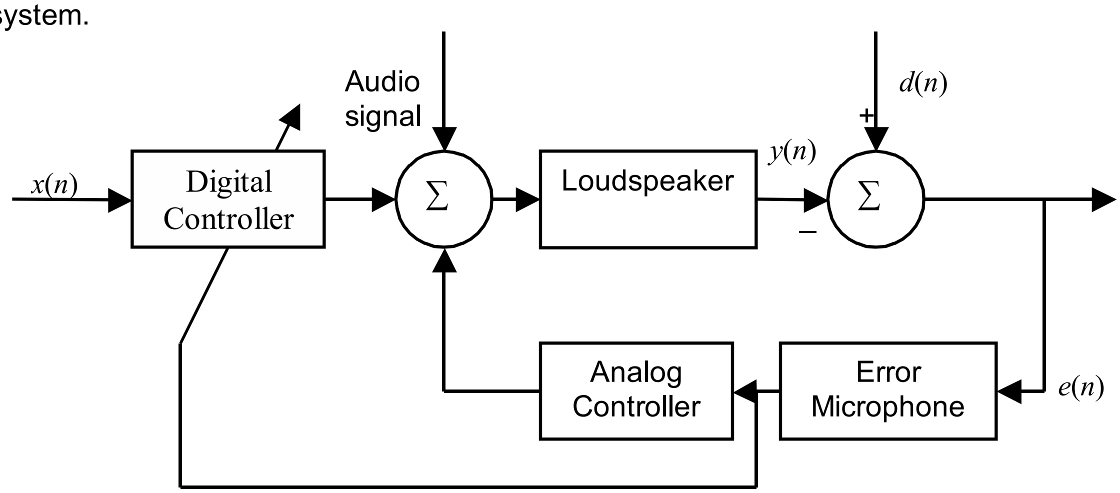

Fig 9. Combined analog and digital control system for headphones ”adapted from [Raf02]”

Practical Applications

Active headphones have been implemented in the laboratory for many years and are now becoming commercially available. Most of these systems use the feedback techniques, and are designed to reduce any external noise, deterministic or random. Headphones operating on the feedforward control principle have also been developed for the selective reduction of periodic noise. Several companies like Bose Corporation, NCT Group, Inc. are developing the active headphones for military use, and commercial applications.

In industrial settings, noise-reducing headphones that use passive, and active attenuation protect the hearing of the workers exposed to deafening levels of sound on a daily basis. In the field of communications, they can enhance the intelligibility of speech. The passengers in aircraft use them for listening to the audio or video flight entertainment. Similar headphones combined with Walkman system provide more comfortable listening to music.

In the aircraft cabin, where the pilot has to hear the speech, and warning sounds, the acitve headphones are much in demand for the cancellation of engine-induced noise. Another application of acitve headphones is its use in audiometry to diagnoise the hearing defects in the human ear, and speech discrimination.

Conclusion and future prospects

The passive and active attenuation methods used in headphones for noise cancelling have been discussed. Also in active cancellation of noise, both the analog and digital models have been described. It is found from the study that both the passive and active attenuation complements each other, good passive attenuation at high frequencies and good active control at low frequencies. Also the digital model is best suited at narrow band frequencies or tonal frequencies whereas the analog model gives good performance for broadband noise cancellation. The best and the optimal solution is to combine all the three characteristic models in a single headphone to use it at a wide range of frequencies.

The development of active noise control systems in the field of acoustics, particularly headphones, has reached a stage where commercial systems are available for protection from noise in a wide variety of applications. The very considerable improvements in the implemenatation of such systems over the past decade or so have been largely due to the availability of poweful and yet relatively cheap DSP devices. Some of the alogirthms like filtered-x LMS alorithm have been tested for headphones on DSP processors [Wan 97]. Experiments have shown drastic improvement in the cancellation of noise in headphones. However for most of these algorithms, the noise cancellation parameters are taken in the absence of audio signal and hence they have to be updated in case of change of noise source or addition of another noise source. There are other alorithms like recursive least squares (RLS) algorithm, which is deterministic in the nature of the noise and hence can be used for real time cancellation of noise without the need for offline updating of its parameters, and Kalman filtering approach, which has a faster rate of convergence. However due to the computational complexity of the these algorithms and the cost involved, there have not been much research on the implementation of these algoithms in noise cancelling headphones. So it will take a few more years to implement these algorithms on the DSP processors to achieve a more better improvement in the cancellation of noise in headphones.

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Suggested Readings

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