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Noise Cancellation In Headphones

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Abstract

Noise cancellation headphones (NCHs) supplement the acoustic isolation characteristic of headphones with active noise reduction. By their nature, headphones block out some degree of external noise because the earcups absorb it, but NCHs go a step further and diminish the noise that manages to get through. This report presents some of the methods used in implementing noise cancellation headphone. The principles of passive attenuation after which active attenuation which includes feedforward and feedback methods, are described.

1. Introduction

The goal of active noise cancellation (ANC) is to reduce the amplitude of the sound pressure level of the noise incident on the receiver or ear by "actively" introducing a secondary, out-of-phase acoustic field, "anti noise". The resulting destructive interference pattern reduces the unwanted sound. Over the past two decades, significant advances in control theory and the development of flexible, programmable, high-speed digital signal processing computers have made it possible to model and implement more complex active noise control systems.

ANC is based on either feedforward control or feedback control. In feedforward control, a reference input coherent with the noise is sensed before it propagates past the secondary source. In feedback control, the active noise controller attempts to cancel the noise without the benefit of an "upstream" reference input. Structures for feedforward ANC are classified into (1) broadband adaptive feedforward control with a control field reference sensor, (2) narrowband adaptive feedforward control with a reference sensor that is not influenced by control field. Feedforward ANC is generally more robust than feedback ANC particularly when the feedforward system has a reference input isolated from the secondary antinoise source.

NCHs rely on the passive acoustic isolation of headphones as well as ANC to provide broadband noise reduction as shown in Fig.1 [Moy01]. Closed-ear headphones can passively block high frequency noise down to about 500Hz. At high frequencies, a well-designed closed-ear headphone can reduce noise by nearly 30dB - the amount of attenuation being dependent upon the quality of the seal from the ear cushion around the ears, the construction of the earcups and the sound-absorbent materials used in the earcups.

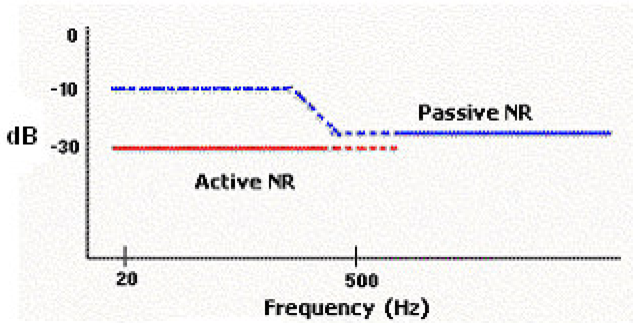

Figure.1 Operational range of active and passive noise reduction, from [Moy01].

2. Passive attenuation

Passive attenuation in an earcup is a function of the front and rear cavity volumes and the driver compliance below the free air resonance. Increased driver compliance can compensate the reduction of passive attenuation if the front cavity is kept small for maximum efficiency [Bos84].

An optimized headphone construction, described by R. Sapiejewski [Sap93] is shown in Fig.2. An ideal cavity is characterized by rigid walls and constant pressure amplitude for wavelengths much larger than the distance across the cavity. The ear cushions should be compliant enough to affect a seal that prevents leaks and has sufficient high density and flow resistance to create such an "ideal" cavity. The diaphragm should be small - less than 1/3 wavelength of the highest audio frequency to be reproduced. There is a trade off between a small front cavity which will minimize the sound pressure required to cancel low frequencies and a large front cavity which offers superior passive attenuation.

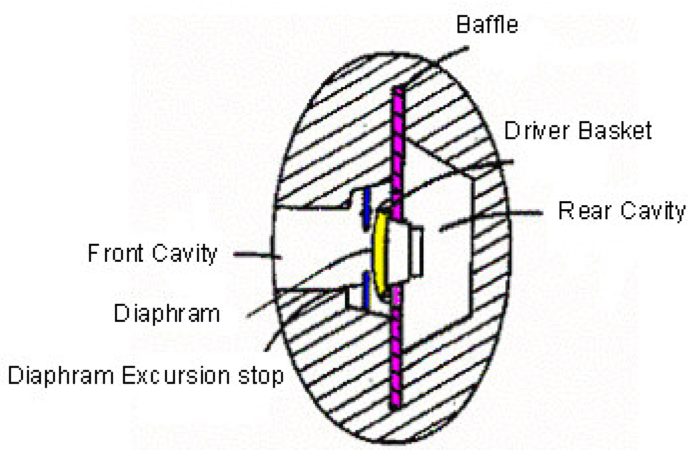

Figure.2 High compliance drive for headphone ANC, as described in [Sap93].

The disadvantage with high compliance driver is its sensitivity to overpressure. The pressure waves due to the movement can pull the voice coil outside the gap or distort the diaphragm when the user moves or takes off the headphones. A tiny barrier or mesh in front of the diaphragm can prevent its distortion. This diaphragm can also be minimized by indenting it so that it quickly recovers when distorted [Sap93].

Where passive attenuation begins to weaken in the low frequencies, ANC takes over. Noise cancellation is less effective at high frequencies due to the limitations of ANC filters and headphone transducers to reproduce accurate high-frequency anti-noise. Not all applications require broadband noise attenuation, and some models of NCH reduce low frequency noise only. A noise reducing headphone should have good attenuation at lower as well as higher frequencies.

3. Feedback control

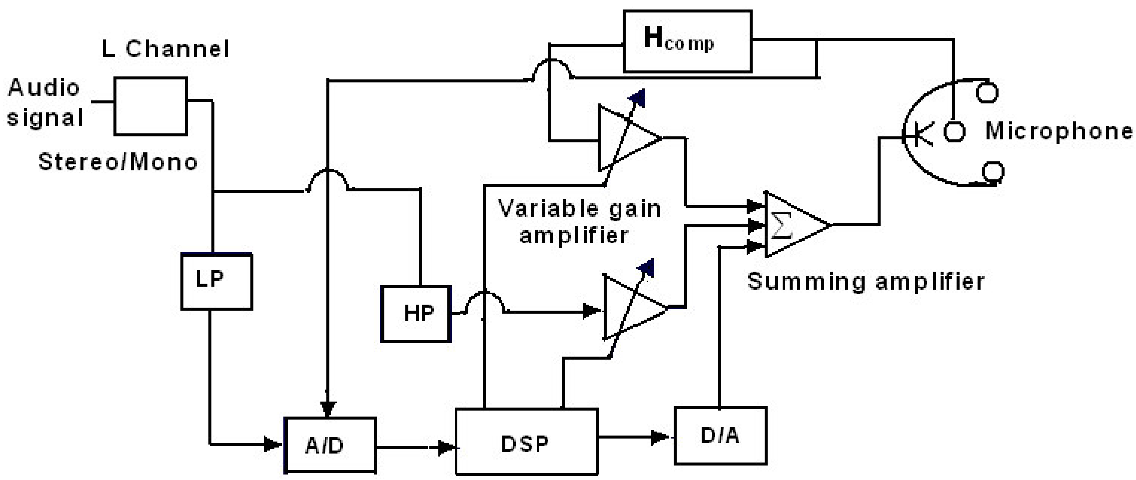

Figure.3 Schematic diagram of noise cancellation circuit for a single channel headset system, as proposed in [McI01].

The active noise cancellation headset system, when an audio signal is given to headphone (for single channel), is as shown in Fig.3, implemented by J.D.McIntosh [McI01].This method can also be used for noise cancellation without audio signal. The essential elements of the system are repeated for the left and right channels like the DSP, analog to digital (A/D) and digital to analog (D/A) converter.

A speaker is mounted within the earcup for receiving and acoustically transducing a signal, combination of noise cancellation signal and audio signal. A microphone is also mounted within the earcup for transducing acoustic pressure within the earcup to a corresponding analog error signal. An analog filter receives the analog error signal and inverts it to generate an analog broadband noise cancellation signal. The analog error signal is also provided to an A/D converter, receives the analog microphone error signal and converts it to a digital error signal. A high pass (HP) and low pass (LP) filters are provided in the audio signal for each channel. The low frequency component of the audio signal is given to DSP via A/D where it is subtracted from the microphone output otherwise acoustic signal cancels the desired audio signal. The high frequency signal is directly given to summing amplifier.

The DSP takes the digital error signal and generates a digital tonal noise cancellation signal using an adaptive digital feedback filter. A digital to analog converter then converts the digital tonal noise cancellation signal to an analog tonal noise cancellation signal so that it can be combined with the analog broadband noise cancellation signal. The resultant composite cancellation signal is provided to the speakers in the earcups to cancel noise within the earcups. The broadband analog cancellation is effective to reduce overall noise within the earcup.

The DSP not only provides active control of the analog cancellation loop gain to maximize the effectiveness of the broadband analog cancellation but also uses the adaptive feedback filter/algorithm to substantially reduce at least the loudest tonal noises penetrating the earcup. The adaptive digital feedback filter may be a regenerative feedback filtered-x least-means-square (FXLMS) algorithm [Kuo96]. This system can be viewed as an “adaptive feedforward system that, in effect, synthesizes or regenerates its own reference signal based only on the adaptive filter output and the error signal” [McI01].

4. Feedforward control

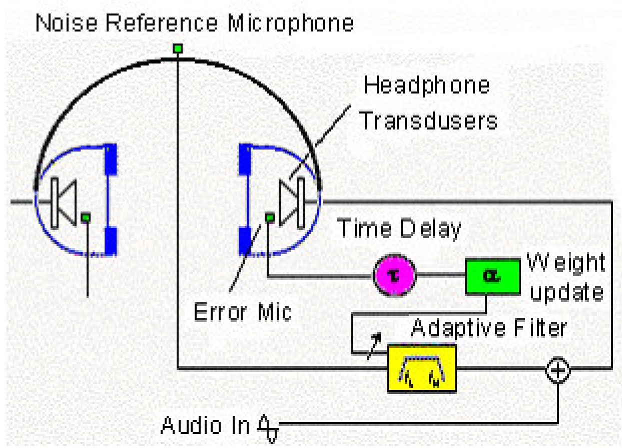

Figure.4 Adaptive Noise cancellation headphone, as described in [Moy01].

Digital filters for noise cancellation are called adaptive filters and can correct for both phase and amplitude errors. A NCH using adaptive noise reduction is shown in Fig.4 was proposed by Chu Moy [Moy01]. A reference microphone at the top of the headband receives a noise signal. The adaptive filter attempts to predict the noise inside the earcup by passing the signal through a transfer function that models the headphone system. The inverse of the predicted noise is added to the desired audio signal and then sent to the headphone transducer. A second microphone inside the earcup measures the resulting sound and generates an error signal to converge the filter to zero for more accurate anti-noise [Moy01].

The algorithm used in an adaptive filter is typically a least means square (LMS) type, which is pre-trained offline (usually with white noise) for broadband operation. The disadvantage to LMS filters is that they must be retrained for changes in the feedback path (e.g., temperature changes, a different person wear the headphones). If the noise is narrow-band or a tone, the performance of the noise cancellation will degrade because LMS filters will try to adapt without converging. For narrow-band noise cancellation, the solution is to use an infinite impulse response (IIR) filter. An IIR is best to model the acoustic system and feedback path, because IIRs have a recursive characteristic to provide an infinite response and have feedforward and feedback sections to generate zeros and poles. A recursive LMS algorithm (made of two LMS filters) can optimally adapt the filter coefficients [Eri87].

The performance of ANC using two sensors is significantly high compared to single sensor particularly the transfer function between canceling speaker and reference sensor was a pure delay of about 40 micro secs [Zan93].

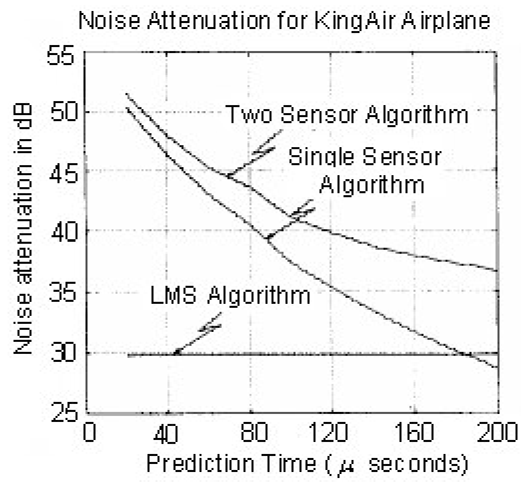

Figure.5 Attenuation obtained using three different algorithms, as described in [Zan93].

5. Adaptive algorithms

Adaptive filtering means that filter parameters such as bandwidth and resonant frequency change with time. This is done by allowing the coefficients of the adaptive filter to vary with time and to be adjusted automatically by an adaptive algorithm. This has the important effect of enabling adaptive filters to be applied in areas where the exact filtering operation required is unknown or nonstationary.

5.1 LMS algorithm

The least mean square (LMS) algorithm is the simplest and is the most universally applicable adaptive algorithm to be used. This algorithm is used for the descending on the performance surface, and is known as the least mean square algorithm. The weights are updated using this algorithm. This algorithm uses a special estimate of the gradient that is valid for the adaptive linear combiner. This algorithm is important because of its simplicity and ease of computation, and it doesn't require off-line gradient estimations or repetitions of data. If the adaptive system is an adaptive linear combiner, and if the input vector x(n) and the desired response d(n) are available at each iteration, the LMS algorithm is generally the best choice for many different applications of adaptive signal processing [Kuo96].

5.2 Filtered- X LMS algorithm

The use of adaptive filter is complicated by the fact that electrical reference signal must be obtained from the acoustic pressure using a microphone. Also, an electrical error signal must be obtained from the residual acoustic noise using as error microphone. Finally, the canceling sound must be produced from the electrical output signal using a loud speaker. Therefore, a number of other transfer functions must be included. The summing junction represents acoustic superposition in the space from the canceling loudspeaker to the error microphone, where the primary noise is combined with the output of the adaptive filter. Therefore it is necessary to compensate for the secondary path transfer function S(z) from y(n) to e(n),which includes the D/A converter, reconstruction filter, power amplifier error microphone etc. [Kuo96].

5.3 Transform Domain Adaptive Algorithm

The main shortage of the filtered –X LMS algorithm, which is widely used in a variety of active noise control (ANC) systems, is that its convergence behavior highly depends on the power spectrum of the filtered reference signal. The more correlated reference signal is the slower convergence speed the adaptive filter can achieve. Transform domain adaptive filter (TDAF) into ANC to solve this problem. To strongly correlate broadband signal, TDAF can achieve much faster convergence speed than LMS with relative low computational cost [Son02]. These three algorithms are elaborated in Appendix A.

6. Practical applications

Noise canceling headphones are particularly useful for workers, operating or working near heavy machinery and engines. The noise is selectively eliminated thus enabling the reception of the desired sounds, such as speech and warning signals.

Cabin noise in small aircraft is a combination of noise from a variety of sources, the major ones being the engine, wind, and propeller. So plane pilots routinely wear noise attenuating headsets to get rid of the unwanted noise. Such headsets usually employ passive noise attenuation in the form of an annular cushion carried on the rim of each earcup. But NCH is proved to be worthy in attenuating noise to a further level which makes the pilot as well as passengers to hear warnings and instructions effectively.

In hospitals to diagnose the hearing defects in human ear and speech discrimination, noiseless room is required which is highly expensive. Most of the hospitals cannot afford to procure such rooms. The noise canceling headphone along with audiometry apparatus can be useful in this application.

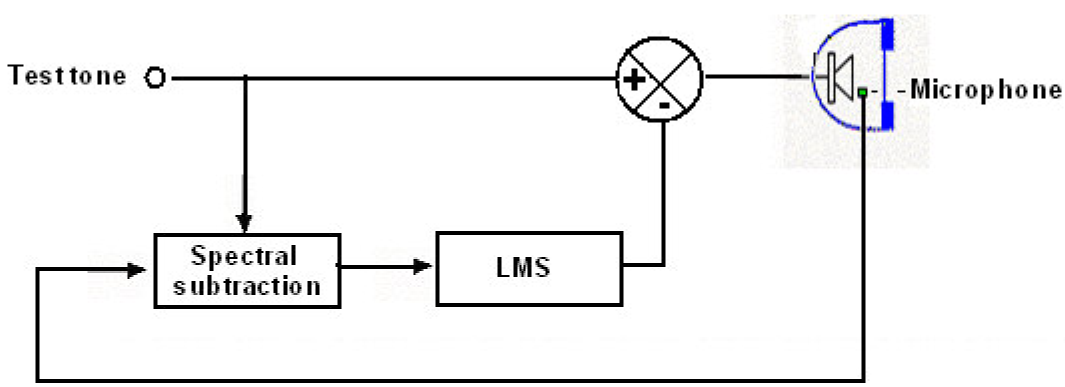

Figure.6 Feedback noise cancellation for tone audiometry.

The proposed NCH for audiometry is as shown in Fig.6. A test tone is given to headphone as input. A microphone is placed inside the headphone which senses noise along with tone. The test tone is subtracted from the microphone output using spectral subtraction before it is given as input to LMS block. The output of the LMS block is an anti noise signal which is added to test tone at the summing block. Finally the speaker gives an output which is a combination of anti noise signal and tone so that anti noise signal cancels the noise and listener only gets test tone. The attenuation required for audiometry application is high compared to other applications of NCHs. So the better passive attenuation as well active attenuation is required.

In military applications, the soldiers usually work noisy environments. NCH are handy in hearing commands through mobile sets devoid of any noise.

As a commercial application NCHs along with personalized music system (Walkman, etc.) can provide better listening conditions for music. Sony, NCT, Aiwa, and Bose Corporation are producing these headphones, mainly used for audio listening. Information of these is given in Appendix B.

7. Conclusion and future prospects

A comprehensive study on passive attenuation and active attenuation methods has been made in this report. The passive attenuation is useful in damping the higher frequencies whereas active attenuation methods are effective in reducing the lower frequencies. Analog compensation is worthy for broadband noise and digital filters are proved to be successful in narrowband cancellation. The combination of above filters covers entire frequency range. The traditional aspects of the active headsets can be further enhanced by improving the design of transducers, passive designs and analog controllers.

Depending upon the application, various adaptive algorithms can be used in digital filters. If speed of convergence is not important LMS algorithm is widely used. For audiometry application either feedforward control or feedback control can be used. Feedfoward adaptive filter is robust considering output of error sensor and reference sensor are uncorrelated and feedback filter is compact.

Active noise reducing headphone has been shown to provide a promising protection from noise using a combination of passive, analog and digital techniques. Future development in DSP could even bring better solutions for more demanding noise environments. The active headset application demonstrated how control and signal processing can be implemented successfully in a single application.

Appendix A

LMS Algorithm

The LMS algorithm is used to create a filter $\boldsymbol{w}$. The input is defined as the vector $\boldsymbol{x}$, the output is the scalar y, and the desired output is the scalar d. The output is calculated according to

$$y(n) = \boldsymbol{x}(n)^T \boldsymbol{w}(n) \tag{1}$$

where the subscript n defines the time indexing. The scalar error e(n) is defined as

$$e(n) = d(n) - y(n) \tag{2}$$

and the filter is updated according to $w(n+1) = w(n) + 2\,\mu\, e(n)\, x(n)$ (3)

where μ is an update coefficient that determines the speed of the adaptation. We also define

$$\boldsymbol{R} = E[\boldsymbol{x}\boldsymbol{x}^T] \tag{4}$$

(where $E[\bullet]$ denotes expectation) as the autocorrelation matrix. The largest valid value of μ is $1/\lambda_{max}$, where λ_{max} is the largest eigenvalue of $\boldsymbol{R}$. Larger values of μ will cause instability and the weights will not converge. The error signal will decrease as the weights converge, and follow a set of exponential curves, each with time constant $\tau_n = 1/(4\mu\lambda_n)$ where λ_n is the n^{th} eigenvalue of $\boldsymbol{R}$.

The LMS algorithm is one of a class of gradient descent algorithms. Another variant, known as the α-LMS algorithm is given by

$$\boldsymbol{w}(n+1) = \boldsymbol{w}(n) + \alpha\, e(n)\boldsymbol{x}(n)/|\boldsymbol{x}(n)|^2 \tag{5}$$

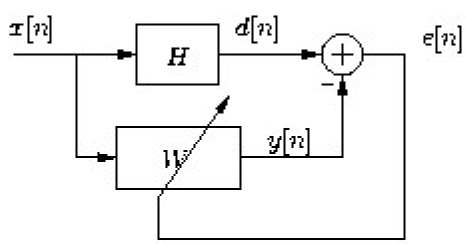

Figure.7 System identification block diagram, from [Kuo96].

Note that 2μ has been replaced by α, and the input signal $\boldsymbol{x}(n)$ is scaled by its magnitude. These changes remove the need for eigenvalue estimation to choose μ. Values in the range $0 < \alpha < 2$ converge, and the time constants are unchanged since the ratio of eigenvalues in $\boldsymbol{R}$ are still the same. This algorithm is particularly useful in the face of nonstationary **x** signals. In this case, the assumption that scaling $\boldsymbol{x}(n)$ leaves the eigenvalue ratio unchanged is false. However, it still relieves us of the estimation of values for μ and thus it is useful for nonstationary $\boldsymbol{x}$.
The main drawback of the LMS algorithm is the speed of convergence becomes very slow if there is a big spread among the eigenvalues of R and changes in the feedback path. The existence of the secondary path affects the performance of ANC filter and the convergence of the LMS algorithm [Kuo96].

Filtered –X LMS algorithm

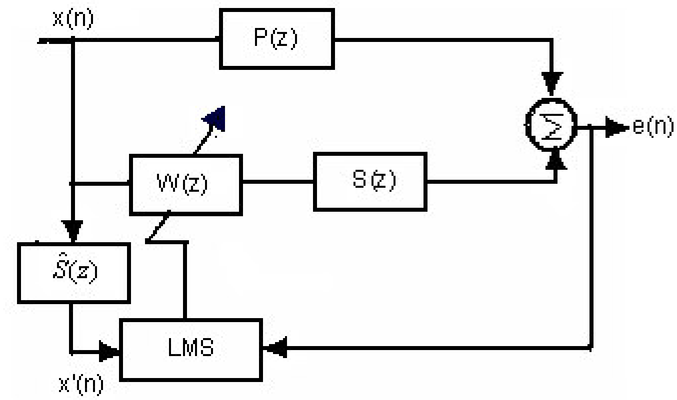

Figure.8 Block diagram of ANC system using the FXLMS algorithm, from [Kuo96].

The compensation can be carried by placing an inverse filter in series with the loudspeaker or placing as identical filter in the reference signal path to the weight update of LMS algorithm.

The placement of the secondary path transfer function following the digital filter W(z) controlled by the LMS algorithm is shown in figure 8.The residual error is expressed as

$$e(n) = d(n) - y'(n) \tag{6}$$

$$= d(n) - s(n) * y(n) \tag{7}$$

$$= d(n - s(n) * [w^T(n)\, x(n)] \tag{8}$$

where s(n) is the impulse response of the secondary path S(z) at time n,* denotes linear convolution,

$$\mathbf{w}(n) = [w_0(n) \quad w_1(n) \ . \ . \ . \qquad w_{L-1}(n)]^T \tag{9}$$

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

$$\mathbf{x}(n) = [x(n)\ x(n-1)\ \ldots \quad x(n-L+1)]^T \tag{10}$$

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,

$$\hat{\xi}(n) = e^2(n) \tag{11}$$

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

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

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 (8)

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

where $\boldsymbol{x}'(n) = [x'(n) \quad x'(n-1) \quad . . . \quad x'(n-L+1]$ (15)

and $\boldsymbol{x}'(n) = s(n) * \boldsymbol{x}(n)$ (16)

Therefore, the gradient estimate becomes

$$\nabla \hat{\xi}(n) = -2\, \mathbf{x}'(n)\, e(n) \tag{17}$$

Substituting equation 17 into 12, we have the FXLMS algorithm

$$W(n) = w(n) + \mu\, \boldsymbol{x}'(n)\, e(n) \tag{18}$$

The implementation of the filtered-x LMS algorithm is 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 [Kuo96].

Transform Domain Adaptive Algorithm

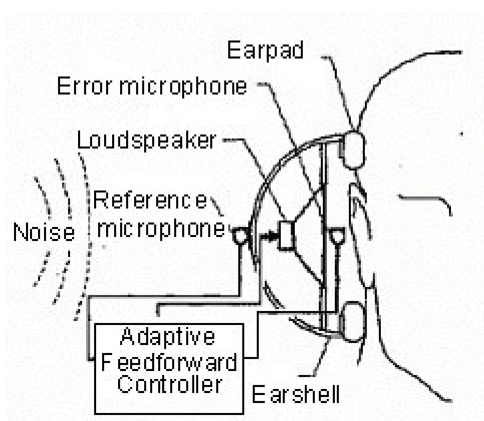

Figure.9 Feedforward ANC headset, from [Son02].

The main difference between Filtered X DCT- LMS and Filtered X- LMS algorithms is that two orthogonal transform parts of Sliding DCT (SDCT) are interposed. The reference signal x(n) is acquired through reference microphone as the error signal e(n) through the error microphone. The primary path P(z) describes the acoustic response from reference microphone to error microphone while the secondary path S(z) consists of transfer functions from adaptive filter output to error microphone .W(z) is the adaptive filter to minimize the residual error signal e(n).To compensate the secondary path effect, $\hat{S}(z)$,the estimate of S(z),is introduced to filter x(n) .

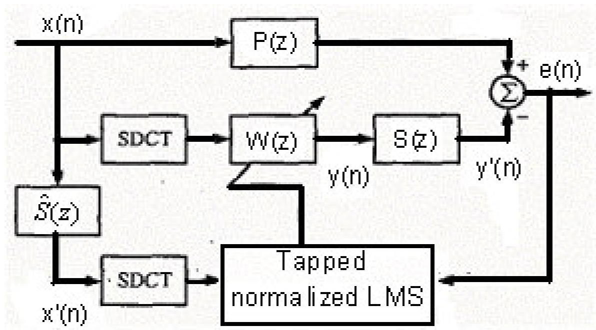

Figure.10 Block diagram of Filtered X DCT-LMS algorithm, as described in [Son02].

The Filtered X DCT-LMS algorithm can provide not only fast, but also uniform convergence speed throughout whole broadband [Son02].

Appendix B

Noise-canceling headphones are offered by several companies a few are listed below. The cost varies from US $69-$269. All companies use the same principle. The noise is reduced by producing anti noise, same magnitude but 180^0 out of phase as noise. This idea dates back to the 19th century, but it took the arrival of silicon processors to provide portable capability for real-time analysis of ambient noise and its cancellation.

With current processors almost all the noise-cancellation is in the low to middle range of human hearing, from about 25 to 1600 Hz. Higher frequencies and sudden sounds are not muted. In practice that means the throb of a bus engine is considerably lessened, while the grinding of subway wheels or the crackle of a gunshot remains unaffected. Claims of 10 decibel attenuation (a 70 percent reduction, according to this measurement's logarithmic scale) in at least part of the hearing range are common. The information regarding the algorithms used and the method in implementing noise canceling headphones is not available.

NCT NoiseBuster

This model is developed by NCT Company. Its cost is US $69. The amount of noise cancellation can be controlled by slide switch. Active noise cancellation is in the range of 20-1500Hz and maximum attenuation is 15 dB between 150-300Hz. Adjustable noise cancellation ranges between 8 - 15 dB.

Brookstone Wireless System

This is a product from Merchandiser Brookstone at US $79. this entry from provides the bulkiest and most isolating of earphones, resembling passive sound mufflers worn by workers on airport runways.

Koss Stereophone

From stereo headphones pioneer Koss, comes this $199. Average noise attenuation is 10dB in the range 30-1000 Hz and the maximum attenuation is 15 dB from 40 to 200 Hz.

Sony MDR-NC20 Headphones

Sony's MDR-NC20 Noise canceling headphones feature the reduction of outside noise by more than 70% (10dB).This headphone is mainly used for listening music against a quieter background using the noise canceller on/off switch.

Sennheiser PXC250

This headphone is from Sennheiser company, cost is $269.The noise attenuation over a broad range of frequencies is around 25 dB upto 1200 Hz.

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