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A Survey on Techniques of Noise Reduction of Audio Signal Using Active Headphones

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ABSTRACT: Noise is defined as any kind of undesirable disturbance, whether it is borne by electrical, acoustic, vibration, or any other kind of media. Noise consists of unwanted waveforms that can interfere with communication. Noise cancellation is a method to reduce or completely cancel out undesirable sound Noise cancellation tries to 'block' the sound at the source instead of trying to prevent the sounds from entering our ear canals. Noise cancellation technology is aimed at reducing unwanted ambient sound, and is implemented through two different methods. The first of these is passive noise cancellation: an approach that focuses on preventing sound waves from reaching the eardrum, and includes devices such as circumaural headphones or earbuds . The other technique used to achieve the same – and often better – result is active noise cancellation, which uses aural overlap and destructive interference to target and attenuate background noise. While passive and active noise cancellation may be applied separately, they are often combined to attain maximum effectiveness in noise cancellation.

KEYWORDS: Ambient Noise Reduction, Active Noise Cancellation Headphones.

I. 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 cancelling 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."

II.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.

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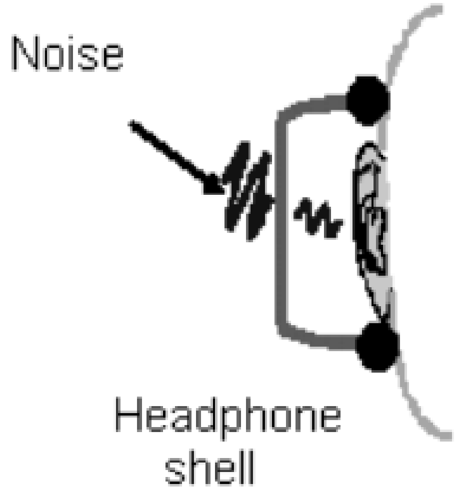

Fig 1.passive noise cancellation

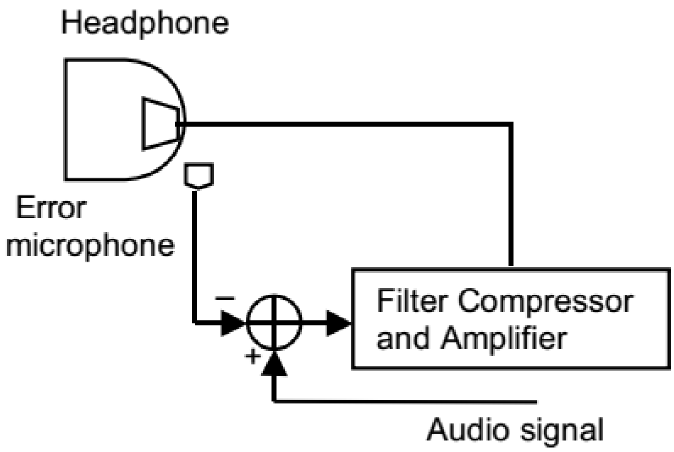

Fig 2 - Noise filtering headphone

or loudspeaker. However the drastic bandpass filtering in these devices does not clean up noise 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 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 antinoise signal may actually introduce noise inside the headphones.

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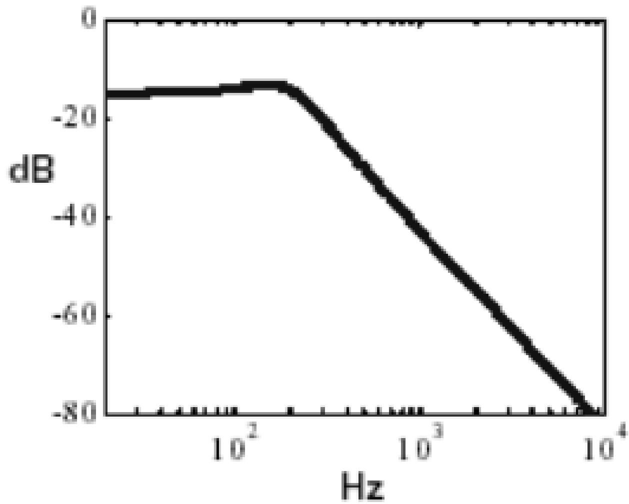

Fig .3 Typical passive attenuation

The passive attenuation of a typical headphone is illustrated in Fig. 3, showing about 15dB of attenuation below the cut-off frequency, and an increasing attenuation above the cut-off 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."

III.ACTIVE ATTENUATION

A.ANALOG FEEDBACK CONTROL

A more accurate anti-noise signal is possible if the microphone is placed inside the ear-cup 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.

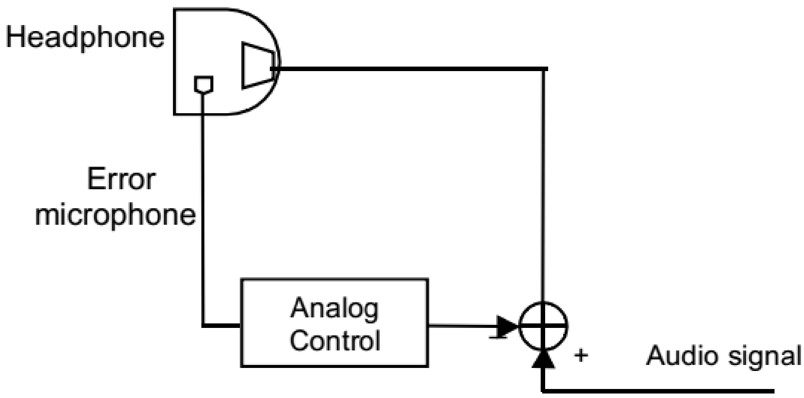

Fig 4. Active headphone attenuation with feedback control

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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 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. 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.

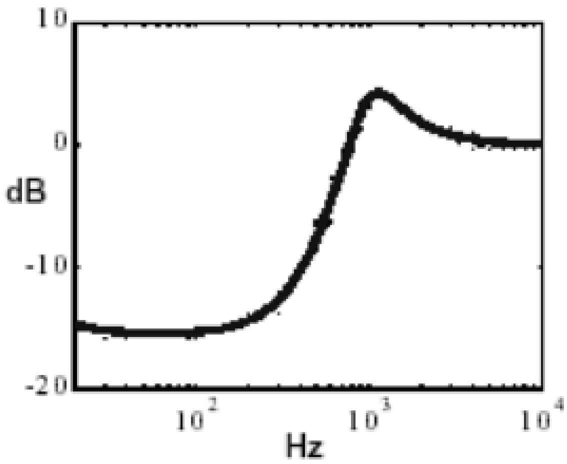

Fig. 5 Typical active attenuation

B.DIGITAL FEED FORWARD CONTROL

Modern active noise cancelling headphones generally use feed forward 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." [May99] 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 feed forward 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 Headphone Error 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. 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.

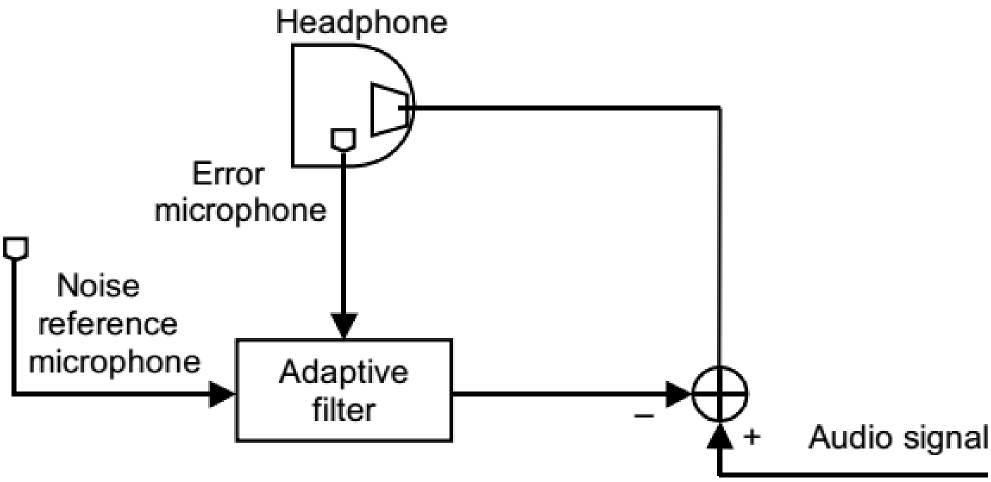

Fig 6 Active headphones with feedforward control

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 band limited or predictable signals can be successfully attenuated. Performance is thus limited to narrow band or tonal noise when using conventional DSP systems.

IV. 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 cancelling headphones are detailed in this section.

A.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 feed forward 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.

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 disadvantage of LMS filters is that they must be retrained for changes in the feedback path (e.g. Temperature changes, a different person wears the headphones).

B.ADJOINT LMS ALGORITHM

The adjoin algorithm was introduced as a low computational alternative to filtered-x LMS algorithm. 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 (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

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equivalent rate of convergence and maladjustment to filtered-x LMS algorithm with a substantial computational savings over Multiple Error LMS algorithm [Wan96].

C.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 feed forward controller W by adding their control outputs at the loudspeaker input as shown in Fig. 7. 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.

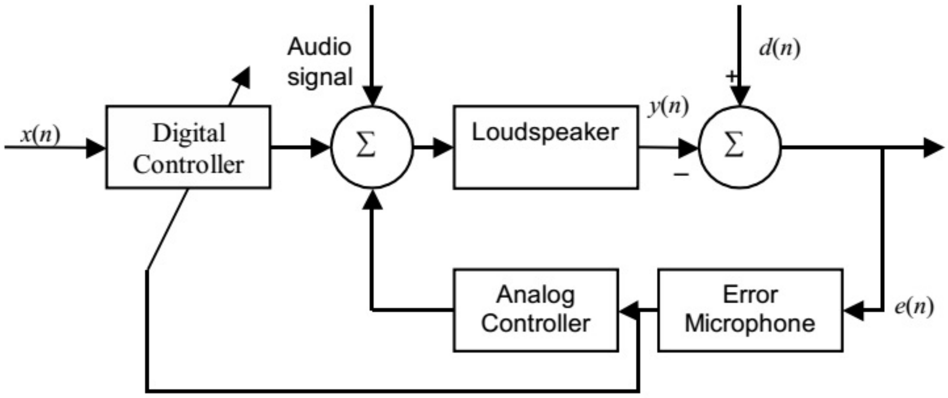

Fig 7. Combined analog and digital control system for headphones

V.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 feed forward 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 active headphones are much in demand for the cancellation of engine-induced noise. Another application of active headphones is its use in audiometry to diagnose the hearing defects in the human ear, and speech discrimination.

VI.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 forprotection from noise in a wide variety of applications. The very

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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 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 better improvement in the cancellation of noise in headphones.

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