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Rapid Expulsion of Acoustic Soft Noise for Noise Free Headphones using RAT

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***Abstract*—Active Noise Cancellation (ANC) is a method of reducing undesired noise in Headphones. ANC is achieved by introducing a canceling “anti-noise” wave through secondary sources, such as from a loud speaker. The traditional approach of acoustic noise control uses passive techniques such as enclosures, barriers, and silencers to attenuate the undesired noise over a broad frequency range; however, they are relatively large, costly, and ineffective at low frequencies. On the other hand, the ANC system efficiently attenuates low-frequency noise where passive methods are either ineffective or tend to be very expensive or bulky. This paper deals about design and Implementation of Rapid Air Technology-Feedback ANC (RAT-FANC) algorithm. The RAT-FANC algorithm is divided into design of ANC, Modified - FANC (MFANC) and then RAT-FANC. The algorithm is implemented using Super Harvard Architecture Computer (SHARC) processor. The implementation of RAT-FANC is based on variations in the signaling paths of actual ANC system to enhance the attenuation range of soft noises for the effective Audio Processing.**

***Index Terms*—ANC, Headphone, MFANC, RAT-FANC, SHARC**

I. Introduction

Sound is a pressure wave, which consists of a compression phase and a rarefaction phase. A noise-cancellation speaker emits a sound wave with the same amplitude but with inverted phase (also known as anti-phase) to the original sound [16]. The waves combine to form a new wave, [13-14] in a process called interference, and effectively cancel each other out - an effect which is called phase cancellation and called Active Noise Control.

Modern active noise control in headphones is generally achieved through the use of analog circuits or digital signal processing. Adaptive algorithms are designed to analyze the waveform of the background aural or non-aural noise, then based on the specific algorithm [10, 12] generate a signal that will either phase shift or invert the polarity of the original signal. This inverted signal (in anti-phase) is then amplified and a transducer creates a sound wave directly proportional to the amplitude of the original waveform, [20] creating in noise-cancellation headphones, speakers may be co-located with the sound source to be attenuated. In this case it must have the same audio power level as the source of unwanted sound [23]. Alternatively, the transducer emitting the cancellation signal may be located at the location where attenuation sounds is wanted (e.g. the user's ear). Noise cancellation at other locations is more difficult as the three dimensional wave fronts [8-11]of the unwanted sound and the cancellation signal could match and create alternating zones of constructive and destructive interference. The interference can be achieved with the help of [9] Adaptive ANC algorithm which generally produces the appropriate power level for the captured external low frequency noise.

The feed forward ANC system is a two way Microphone system consists of two Microphones namely, reference Mic and error Mic in which the reference Mic is correlated with the error Mic [12-15]. With the feed-forward approach, if the acoustic isolation between the speaker and the feed-forward microphone is good enough, there will be no influence on the playback path [17]. The general Feed Forward Active Noise Control System [18-19] is shown in Fig.1.1. With a feed-forward topology, the designer analyses the acoustics of the headset to determine how the noise is affected by the time it reaches the ear, in terms of frequency, phase, and amplitude [22-25]. This transfer function, G(w), is then modeled electrically and inserted between the microphone and speaker. Feed-forward designs can be subject to directional issues, so the microphone must be Omni directional [20]. Also, the noise channel can’t be mechanically concentrated [21]. Because the microphone must acquire the noise before it gets to the ear, parallel acoustic paths must be minimized [20-22].

The biggest challenge with feed-forward solutions is ensuring the controlled environment around the user ears [9]. Different users will have different ear shapes and sizes, and their headsets will fit differently [17-18].

The Closed loop systems generally have minimum resolution; speed and reserved memory can be made efficient Processing of soft noises using Microcontrollers [10]. For narrow-band noise cancellation, the solution is to use an Infinite Impulse Response (IIR) filter because it is best to model the acoustic system and feedback path, because IIRs have a recursive characteristic to provide an infinite response.

It has feed forward and feedback sections to generate zeros and poles. This produces the instability in the system. This proves that the closed loop analysis of the Feedback ANC is quiet inefficient and hence weight updating and use of stable recursive filter is needed.

The Feed forward ANC systems are Omni directional and it provides different values of attenuation. The change in attenuation values result in Poor Active Noise Cancellation.

The Closed loop Feedback ANC systems are processed with Microcontrollers with reserved speed and memory. This leads to restriction of intelligent sound. So, a system with high SNR ratio with stable attenuation range without restricting intelligent sound has to be developed.

The basic solution for the above problems is the design and Implementation of FANC algorithm is shown in Fig.1 .The inversion operation of the LMS path is the key feature for analyzing the stability of the system. The stability is found to be worse since because of the lack of appropriate control paths in the secondary path. The system generally focused on the estimation of the secondary paths but not much about the signaling path for the every scheme of blocks in the system. The property of digital signal processing, canceling the noise at the frequency f_M, which generally requires the data sampling rate $f_S \geq 2f_M$. Be-cause the real-time processing load of ANC includes numerous multiply-accumulate operations to implement the adaptive filter and LMS algorithm during one Some researchers have developed ANC systems for headsets to reduce environmental noise while listening to music or working in noisy environments [9]–[11].

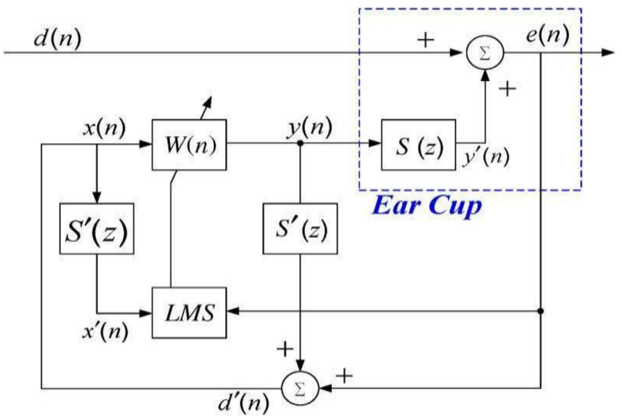

Fig. 1. Block diagram of adaptive FANC in a headset.

These systems include adaptive feedback active noise control (FANC). Fig. 1 presents the main concept on which a headset with the FANC is based. One microphone in the ear cup measures unwanted noise. Anti-noise, which has the same magnitude but is phase-shifted by 180° from the unwanted noise, is generated using the FANC algorithm to cancel the undesired noise. To implement FANC, in [9], an integrated approach to producing an ANC headset has been proposed for audio and communication applications. It emphasizes algorithmic development and verification of effectiveness of such integrated headsets. Simulations and experiments have been conducted to study performance.

The ideal position of the error microphone, the training signal used the selection of adaptive algorithms, and the structures have all been addressed in [10]. Measurements of a DSP have also been made in real time to evaluate and verify performance. Song [11] proposed a means of adding an analog feedback loop to a digital ANC system. This hybrid system can be used to reduce disturbances during the identification of the secondary path, such as when the user adjusts the ear cup, producing a robust system.

II. Modified FANC Headset Based on a Microcontroller

The block diagram of the Active Noise Control is shown in the Fig. 2 and it replaces the Reference Microphone by its Feedback path and hence it is also called as Feedback ANC system where the Signal vectors are denoting the flow of the path. In the ANC system, a Transfer function which is a Band pass filter is included in the Primary path. The general ANC system deals only the Noise attenuation and thus the Noise source is provided in the Primary path. The secondary path is the path which exists between the error microphone and the loudspeaker. The secondary path is meant for the anti-noise generation.

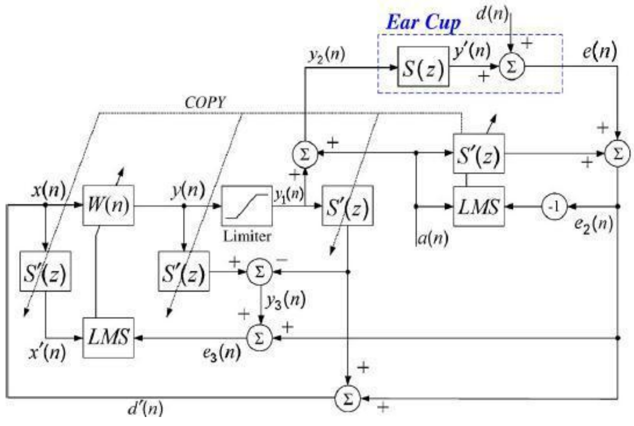

Fig. 2. Block diagram of Modified FANC in a headset.

The ANC will be performed at this stage and there exist some feeble noise elements which are then subjected to process again and again. In this system, $S(z)$ is the transfer function of the secondary path in the ear cup and $S'(z)$ is the estimate of the secondary path. The Lth-order adaptive finite-impulse response (FIR) filter $W(z)$ uses LMS algorithm to update its coefficients. It is based on the theory of adaptive control. The primary noise, anti-noise, and residual noise are denoted $d(n)$, $y'(n)$, and $e(n)$, respectively. Since the primary noise $d(n)$ is canceled by the anti-noise $y'(n)$, it is unavailable during the noise cancellation processes. The typical disadvantages - processing speed of MCU is about 40MHz. Only Basic operations like +/-/*/ and fixed point operations can be performed. The memory space is about 2K without any external interface. Very lower resolution. There is no consideration about the floating point operations.

III. RAT-FANC ALGORITHM

Noise cancellation at other locations is more difficult as the three dimensional wave fronts of the unwanted sound and the cancellation signal could match and create alternating zones of constructive and destructive interference. In small enclosed spaces (e.g. the passenger compartment of a car) such global cancellation can be achieved via multiple speakers and feedback microphones, and measurement of the modal responses of the enclosure and this creates change in signalling path. This technology is called RAT (Rapid Air Technology) and shown in Fig.3.

In this system, F(z) is the transfer function of the secondary path in the ear cup. The Lth-order adaptive finite-impulse response (FIR) filter, W(z) uses LMS algorithm to update its coefficients. The Music signal, primary noise, anti-noise, and residual noise are denoted v(n), $e_o(n)$, $y_2(n)$, and y(n), respectively. Since the primary noise $e_o(n)$ is canceled by the anti-noise $y_2(n)$, it is unavailable during the noise cancellation processes. Therefore, the primary noise that must be estimated before the LMS algorithm can be implemented in the FANC system.

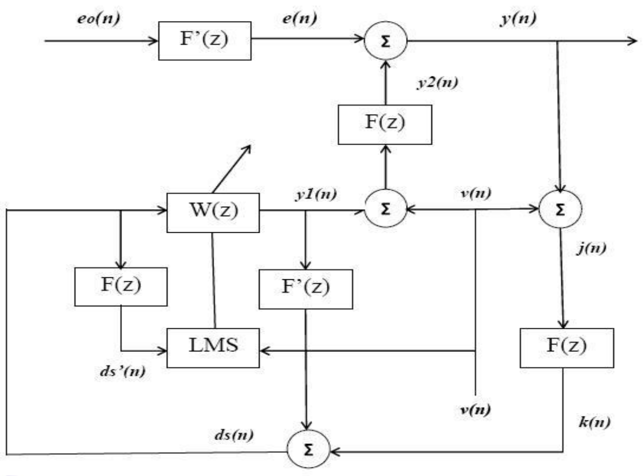

Fig. 3. Block diagram of RAT-FANC in a headset.

The secondary signal of this system, $y_2(n)$ is given as,

$$\mathbf{y_2(n)= y_1(n)+v(n)} \quad \ldots\ldots(1)$$

where,

$$\mathbf{y_1(n)= w^T(n-1)*ds(n)} \quad \ldots\ldots(2)$$

The coefficients of the signal vectors of the adaptive filtering system can be estimated as,

$$\mathbf{w(n)= T\,[w_0(n)\ w_1(n)\ w_2(n)\ldots\ldots w_{L-1}(n)]} \quad \ldots\ldots(3)$$

$$\mathbf{ds(n)= T[ds(n)\ ds(n-1)\ ds(n-2)\ldots\ldots ds(n-L+1)]} \quad \ldots(4)$$

The instantaneous squared error gets minimized by the adaptive filtering using FxLMS algorithm,

$$\mathbf{w(n+1)=w(n)+\mu*ds'(n)v(n)} \quad \ldots\ldots(5)$$

The filtered signal vectors, ds‟(n) and the primary noise will not be available in the ANC operation and hence estimation of the desired signal is needed and it will be used also as a Reference signal and in general, the secondary path estimate is used to synthesis the reference signal,

$$\mathbf{ds'(n)= f(n)*ds(n)} \quad \ldots\ldots(6)$$

$$\mathbf{ds(n)= e(n)+k(n)} \quad \ldots\ldots(7)$$

$$\mathbf{ds(n)= e(n)+(j(n)*f(n))} \quad \ldots\ldots(8)$$

$$\mathbf{ds(n) = e(n)+ \Sigma\, f(n).j(n-m-1)} \quad \ldots\ldots(9)$$

The LMS filtering is proposed and the LMS block can also be implemented using 5 different ways. The Input can be a sample based scalar, Music or a single channel frame based signal and the desired signal must have the same data type, frame status, complexity and dimensions as like the input signal. The output should have the same frame status as i/p and the error , itself the result of subtracting the o/p from the desired signal. We select the LMS for the algorithm parameter; the block calculates the filter weights using the Least Mean Square Algorithm. The Algorithm can be defined as,

$$\mathbf{e(n)=e_0(n)-y_2(n)} \quad \ldots\ldots(10)$$

$$\mathbf{w(n)= w(n-1)+f(ds'(n),v(n),\mu)} \quad \ldots\ldots(11)$$

The Weight update function for the LMS adaptive filter algorithm is,

$$\mathbf{f(ds'(n),v(n), \mu) = \mu*v(n)ds*(n)} \quad \ldots\ldots(12)$$

where,

ds‟(n) – Vector of buffered i/p samples at step „n‟

ds*(n)- Complex Conjugate of vector of buffered input samples w(n)- Vector of filter weight estimates at step „n"

μ- Adaptation step size

When we select the Normalized LMS Algorithm parameter, the block calculates the filter weights using normalized LMS algorithm. The weight update function for the normalized LMS algorithm is defined as,

f(ds'(n),v(n),μ)= μ*ds'*(n)v(n) ……………….………..(13)

f(ds'(n),v(n),μ)= μds'*(n)v(n)/(Ɛ+ds'(n)ds^H(n)) ……(14)

To overcome the potential numerical instability in the update of weights, a positive constant epsilon is added in denominator. Use of filter length parameter to specify the length of the filter weights vector

w(n)=(1-μα)w(n-1)+f(ds'(n),v(n),μ) …………………..(15)

where

(1-μα)- Leakage factor; 0<**(1-μα)<1**

The residual error does not significantly harm the performance of the active noise control system during its operation in the chosen task. The above plot with the sampling frequency of 8 KHz and filter length of 360 but the attenuation level of the MFANC system is of about 35dB. The MFANC system increased the suppression of noise for few extents but the design of such system produces stable output for any noise level whereas the attenuation will be minimal in range.

Power Level Estimation: Motor Noise		
Original Noise Power (dB)	**Attenuation (dB)**	
	FANC System	**MFANC System**
60	**25**	**35**
120	**15**	**35**
185	**12**	**35**

IV. Experimental Results

The output of MFANC gives the estimation of Original Noise vs Anti-noise generated by the system which increases in the attenuation for nearly certain limit. The Original noise which is shown in Fig.4 and the graph will provide the overall strength and the Anti-noise which is generated will have the nearer strength equal to the Original error and the coefficients of both the true and estimated path. Only the tail of the true impulse response is not estimated accurately. Coefficients are important in Filter design and as well as in Noise Cancellation

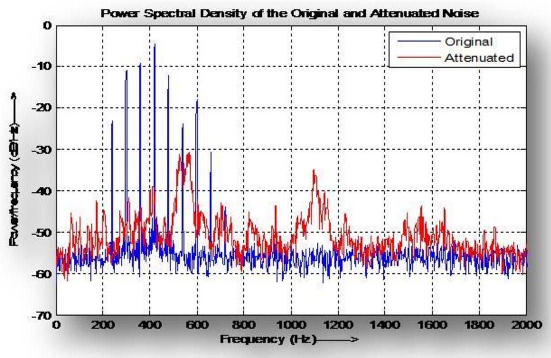

Fig. 4. PSD of Original and Attenuated Noise using MFANC in a headset.

Table 1 Original Noise Attenuation in FANC vs MFANC systems

The above table shows that the performance of FANC systems are degrading whenever there is any increase in the original noise level but the MFANC systems will provide the stability in each power level. On the same hand, the performance of an ANC system can be enhanced by RAT Algorithm and the result is shown in Fig.5

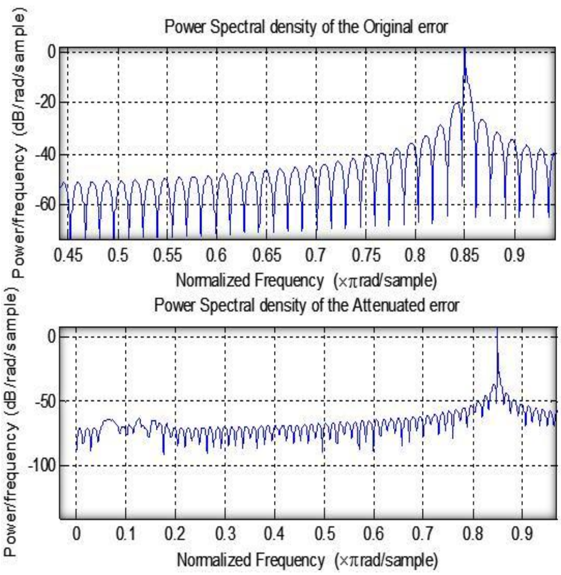

Fig. 5. PSD of Original and Attenuated Noise using RAT-FANC in a headset.

The attenuation range of Noise using RAT-FANC systems is better in nature when compared with the existing FANC and MFANC systems. The noise control can be done up to 50 dB and the performance of each systems are shown in the following table as,

Table 2 Noise Attenuation of various Noises using RAT-FANC Systems

Power Level Estimation: Motor Noise			
Power level (dB)	Attenuation Range(dB)		
	FANC System	MFANC System	RAT-FANC System
60	25	35	47
121	15	35	48
183	10	35	47

Table 3 Original Noise Attenuation in RAT-FANC systems

The RAT algorithm primarily based on the uses the change in the signaling path to LMS filter. This feature enhanced the effective noise cancellation in an audio environment. The comparative statement is given as that the RAT-FANC Algorithm provides the noise cancellation of about 48 dB. The hardware implementation is mainly using SHARC Processor.

V. Conclusion

This paper provides the effective solution for the active noise control over any other existing audio headsets. The RAT algorithm primarily based on the uses the change in the signaling path to LMS filter. This feature enhances the effective noise cancellation in an audio environment and later brings the unbelievable technological arena in the field of Mobile communication and in Audio Engineering.

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Different Noises	Noise Range(dB)	Attenuation Range (dB)
Fan Noise	90	48
FM Noise	75	50
AM Noise	60	50
Background Noise	70	48

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