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ADAPTIVE NOISE CANCELLATION FOR AUDIO SIGNAL PROCESSING

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ABSTRACT

In any communication system noise always has been a major area of concern. Noise signals affect the transmitted signals during transmission and hence noise cancellation of audio signal is key challenge in Audio Signal Processing. Every signal that we acquire at the receiver end of any electronic communication system is affected by noise. So for de-noising of these signals there are many schemes for noise cancellation but most effective scheme to accomplish noise cancellation is to use adaptive filter. Active Noise Cancellation (ANC) is achieved by introducing "antinoise" wave through an appropriate array of secondary sources. These secondary sources are interconnected through an electronic system using a specific signal processing algorithm for the particular cancellation scheme. Adaptive Noise Cancellation uses two inputs - primary and reference. The primary input receives signal from the signal source which has been corrupted with a noise uncorrelated to the signal. The reference input receives noise signal uncorrelated with the signal but correlated in some way to the noise signal in primary input. The reference input is adaptively filtered to obtain a close estimate of primary input noise which is then subtracted from the corrupted signal at the primary input to produce an estimate of a clean uncorrupted signal, which is the Adaptive Noise Cancellation output. A desired signal corrupted by noise can be recovered by feeding back this output to Adaptive Filter and implementing Least Mean Square algorithm to minimize output power. The audio signal corrupted with noise is used as a primary input and a noise signal is used as reference input. This approach provides higher quality of noise cancellation hence improving signal to noise ratio (SNR) and minimizes mean square error (MSE) there by producing high quality signal for any signal processing application.

INDEX TERMS: *Audio signal processing, Noise cancellation, Adaptive filters.*

I. INTRODUCTION

Noise can be defined as an unwanted signal that interferes with the communication or measurement of another signal. A noise itself is an information-bearing signal that conveys information regarding the sources of the noise and the environment in which it propagates.

The types and sources of noise and distortions are many and include:

- **Acoustic noise:** emanates from moving, vibrating, or colliding sources and is the most familiar type of noise present in various degrees in everyday environments. Acoustic noise is generated by such sources as moving cars, air-conditioners, computer fans, traffic, people talking in the background, wind, rain, etc.
- **Electromagnetic noise:** present at all frequencies and in particular at the radio frequencies. All electric devices, such as radio and television transmitters and receivers, generate electromagnetic noise.
- **Electrostatic noise:** generated by the presence of a voltage with or without current flow. Fluorescent lighting is one of the more common sources of electrostatic noise.
- **Processing noise:** the noise that results from the digital/analog processing of signals, e.g. quantization noise in digital coding of speech or image signals, or lost data packets in digital data communication systems.

Depending on its frequency or time characteristics, a noise process can be classified into one of several categories as follows:

- **Narrowband noise:** a noise process with a narrow bandwidth such as a 50/60 Hz 'hum' from the electricity supply.
- **White noise:** purely random noise that has a flat power spectrum. White noise theoretically contains all frequencies in equal intensity.
- **Band-limited white noise:** a noise with a flat spectrum and a limited bandwidth that usually covers the limited spectrum of the device or the signal of interest.
- **Coloured noise:** non-white noise or any wideband noise whose spectrum has a non-flat shape; examples are pink noise, brown noise and autoregressive noise.
- **Impulsive noise:** consists of short-duration pulses of random amplitude and random duration.
- **Transient noise pulses:** consists of relatively long duration noise pulses.

II. AUDIO SIGNAL PROCESSING

An audio signal is a representation of sound, typically as an electrical voltage. Audio signals have frequencies in the audio frequency range of roughly 20 to 20KHz (the limits of human hearing). Audio signals may be synthesized directly, or may originate at a transducer such as a microphone, musical instrument pickup, phonograph cartridge, or tape head. Loudspeakers or headphones convert an electrical audio signal into sound. Digital representations of audio signals exist in a variety of formats.

Signal distortion is the term often used to describe a systematic undesirable change in a signal and refers to changes in a signal due to the non–ideal characteristics of the transmission channel, reverberations, echo and missing samples. Noise and distortion are the main limiting factors in communication and measurement systems. Therefore the modeling and removal of the effects of noise and distortion have been at the core of the theory and practice of communications and signal processing. Noise reduction and distortion removal are important problems in applications such as cellular mobile communication, speech recognition, image processing, medical signal processing, radar, sonar, and in any application where the signals cannot be isolated from noise and distortion.

The usual method of estimating a signal corrupted by additive noise is to pass it through a filter that tends to suppress the noise while leaving the signal relatively unchanged i.e. direct filtering.

The design of such filters is the domain of optimal filtering, which originated with the pioneering work of Wiener and was extended and enhanced by Kalman, Bucy and others.

Filters used for direct filtering can be either Fixed or Adaptive.

Fixed filters - The design of fixed filters requires a priori knowledge of both the signal and the noise, i.e. if we know the signal and noise before hand, we can design a filter that passes frequencies contained in the signal and rejects the frequency band occupied by the noise.

Adaptive filters - Adaptive filters, on the other hand, have the ability to adjust their impulse response to filter out the correlated signal in the input. They require little or no a priori knowledge of the signal and noise characteristics.(If the signal is narrowband and noise broadband, which is usually the case, or vice versa, no a priori information is needed; otherwise they require a signal(desired response) that is correlated in some sense to the signal to be estimated.) Moreover adaptive filters have the capability of adaptively tracking the signal under non-stationary conditions.

Noise Cancellation is a variation of optimal filtering that involves producing an estimate of the noise by filtering the reference input and then subtracting this noise estimate from the primary input containing both signal and noise.

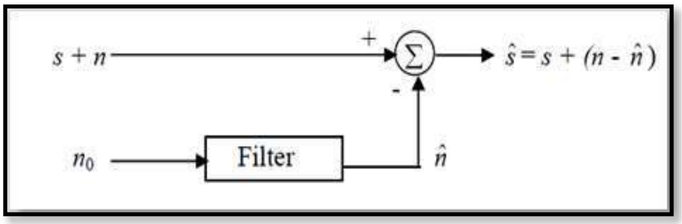

Fig. 1 Noise cancellation using a filter

It makes use of an auxiliary or reference input which contains a correlated estimate of the noise to be cancelled. The reference can be obtained by placing one or more sensors in the noise field where the signal is absent or its strength is weak enough. Subtracting noise from a received signal involves the risk of distorting the signal and if done improperly, it may lead to an increase in the noise level. This requires that the noise estimate should be an exact replica of n. If it were possible to know the relationship between n and nˆ , or the characteristics of the channels transmitting noise from the noise source to the primary and reference inputs are known, it would be possible to make nˆ a close estimate of n by designing a fixed filter. However, since the characteristics of the transmission paths are not known and are unpredictable, filtering and subtraction are controlled by an adaptive process. Hence an adaptive filter is used that is capable of adjusting its impulse response to minimize an error signal, which is dependent on the filter output. The adjustment of the filter weights, and hence the impulse response, is governed by an adaptive algorithm. With adaptive control, noise reduction can be accomplished with little risk of distorting the signal. Infact, Adaptive Noise Canceling makes possible attainment of noise rejection levels that are difficult or impossible to achieve by direct filtering.

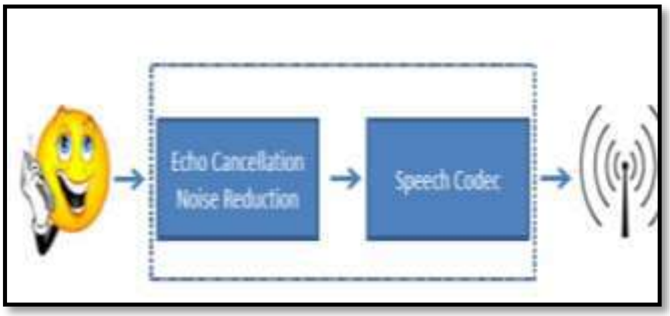

Fig. 2 Illustration of voice call flow in mobile

III. ADAPTIVE FILTER

An adaptive filter is a filter that self-adjusts its transfer function according to an optimizing algorithm. Because of the complexity of the optimizing algorithms, most adaptive filters are digital filters that perform digital signal processing and adapt their performance based on the input signal. In contrast, a non-adaptive filter has static filter coefficients, which together form the transfer function .Since for the desired processing operation some parameters (for instance, the properties of some noise signal) are not known in advance for some application adaptive coefficients are required. In this case it is common to use an adaptive filter, which uses feedback to modify the values of the filter coefficients and thus its frequency response. The adaptive process involves the use of a cost function, which is a criterion for optimum performance of the filter (for example, minimizing the noise component of the input), to feed an algorithm, which determines how to modify the filter coefficients to minimize the cost on the next iteration.

In speech communication from a noisy acoustic environment such as a moving car or train, or over a noisy telephone channel, the speech signal is observed in an additive random noise. In signal measurement systems the information-bearing signal is often contaminated by noise from its surrounding environment. The noisy observation y(m) can be modeled as

$y(m)=x(m)+n(m)$

where x(m) and n(m) are the signal and the noise, and m is the discrete-time index.

In some situations, for example when using a mobile telephone in a moving car, or when using a radio communication device in an aircraft cockpit, it may be possible to measure and estimate the instantaneous amplitude of the ambient noise using a directional microphone. The signal x(m) may then be recovered by subtraction of an estimate of the noise from the noisy signal.

Fig. 3 shows a two-input adaptive noise cancellation system for enhancement of noisy speech. In this system a directional microphone takes as input the noisy signal x(m) + n(m), and a second directional microphone, positioned some distance away, measures the noise α n(m+τ).

The attenuation factor α and the time delay τ provide a rather over-simplified model of the effects of propagation of the noise to different positions in the space where the microphones are placed. The noise from the second microphone is processed by an adaptive digital filter to make it equal to the noise contaminating the speech signal, and then subtracted from the noisy signal to cancel out the noise.

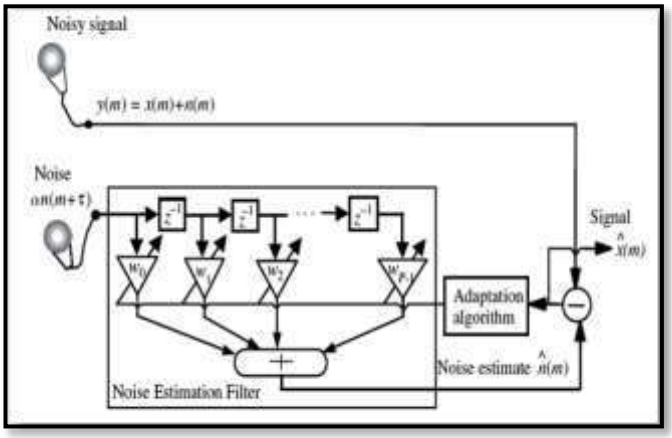

Fig. 3 A two-microphone adaptive

As their own name suggests, adaptive filters are filters with the ability of adaptation to an unknown environment. This family of filters has been widely applied because of its versatility (capable of operating in an unknown system) and low cost (hardware cost of implementation, compared with the non-adaptive filters, acting in the same system).

The ability of operating in an unknown environment added to the capability of tracking time variations of input statistics makes the adaptive filter a powerful device for signal-processing and control applications. Indeed, adaptive filters can be used in numerous applications and they have been successfully utilized over the years . As it was before mentioned, the applications of adaptive filters are numerous. For that reason, applications are separated in four basic classes: identification, inverse modeling, prediction and interference cancelling.

All the applications above mentioned, have a common characteristic: an input signal is received for the adaptive filter and compared with a desired response, generating an error. That error is then used to modify the adjustable coefficients of the filter, generally called weight, in order to minimize the error and, in some optimal sense, to make that error being optimized, in some cases tending to zero, and in another tending to a desired signal.

An adaptive filter is defined by four aspects:

- The signals being processed by the filter
- The structure that defines how the output signal of the filter is computed from its input signal
- The parameters within this structure that can be iteratively changed to alter the filter's input-output relationship
- The adaptive algorithm that describes how the parameters are adjusted from one time instant to the next

IV. ADAPTIVE ALGORITHMS

Adaptive algorithms are useful for adaptation of digital filters. The conventional adaptive algorithms (RLS, LMS, NLMS) are analyzed in this section.

A. Recursive Least Square Algorithm (RLS)

The recursive least square (RLS) algorithm [4, 5] was proposed in order to provide superior performance compared to those of the LMS algorithm and its variants [6, 7], with few parameters to be predefined, especially in highly correlated environments. In the RLS algorithm, an estimate of the autocorrelation matrix is used to decorrelate the current input data.

Even though the RLS algorithm has very good performance in such environments, it actually suffers from its high computational complexity. Also, in RLS algorithm, the forgetting factor λ has to be chosen carefully such that its value should be very close to one in order to ensure stability and convergence of the RLS algorithm.

$$w(n) = w(n-1) + k(n)e(n)$$

$$k(n) = \frac{\lambda^{-1}\Phi_{\lambda}^{-1}(n-1)x(n)}{1+\lambda^{-1}x^{T}(n)\Phi_{\lambda}^{-1}(n-1)x(n)}$$

where, x(n) is the input vector of time delayed input values, w(n) represents the coefficients of the adaptive FIR filter tap weight vector at time n and k(n) is a function of forgetting factor correlation matrix $\phi(n)$.

However, this in turn poses a limitation for the use of the algorithm because small values of λ may be required for signal tracking if the environment is non-stationary [8].

B. Least-Mean-Square Algorithm (LMS)

The LMS algorithm [9], is a stochastic gradient-based algorithms as it utilizes the gradient vector of the filter tap weights to converge on the optimal wiener solution. With each iteration of the LMS algorithm, the filter tap weights of the adaptive filter are updated according to the following formula:

$$w(n+1) = w(n) + 2\mu e(n)x(n)$$

where, x(n) is the input vector of time delayed input values, w(n) represents the coefficients of the adaptive FIR filter tap weight vector at time n and μ is known as the step size.

Selection of a suitable value for μ is imperative to the performance of the LMS algorithm, if the value μ is too small, the time adaptive filter takes to converge on the optimal solution will be too long; if μ is too large the adaptive filter becomes unstable and its output diverges.

C. Normalized Least-Mean-Square Algorithm

In the standard LMS algorithm, when the convergence factor μ is large, the algorithm experiences a gradient noise amplification problem. This difficulty is solved by NLMS (Normalized Least Mean Square) algorithm. The correction applied to the weight vector w(n) at iteration n+1 is "normalized" with respect to the squared Euclidian norm of the input vector x(n) at iteration n.

The NLMS algorithm can be viewed as a time-varying step-size algorithm, calculating the convergence factor μ as follows.

$$\mu(n) = \frac{\alpha}{c + \|x(n)\|^2}$$

where α is the NLMS adaption constant, which optimize the convergence rate of the algorithm and should satisfy the condition 0<α<2, and c is the constant term for normalization, which is always less than 1.

V. ACTIVE NOISE CANCELLING

The active noise cancelling, also called adaptive noise cancelling or active noise canceller belongs to the interference cancelling class. The aim of this algorithm, as the aim of any adaptive filter, is to minimize the noise interference or, in an optimum situation, cancel that perturbation.

The approach adopted in the ANC algorithm, is to try to imitate the original signal x(n). A scheme of the ANC can be viewed in Fig.4. In the ANC, as explained before, the aim is to minimize the noise interference that corrupts the original input signal. In the figure above, the desired signal d(n) is composed by an unknown signal, that s(n) is called corrupted for an additional noise n2(n), generated for the interference. The adaptive filter is then installed in a place that the only input is the interference signal n1(n).

The signals n1(n) and n2(n) are correlated. The output of the filter y(n) is compared with the desired signal d(n), generating an error e(n). That error, which is the system output, is used to adjust the variable weights of the adaptive filter in order to minimize the noise interference. In an optimal situation, the output of the system e(n) is composed by the signal s(n), free of the noise interference n2(n). In the ANC, as explained before, the aim is to minimize the noise interference that corrupts the original input signal.

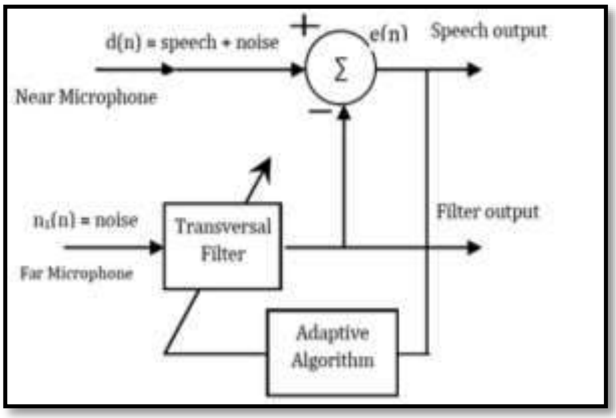

Fig. 4 Adaptive Noise Canceller (ANC)

VI. PERFORMANCE MEASURES IN ADAPTIVE SYSTEMS

Some important measures will be discussed in the following:

A. Convergence Rate

The Convergence rate determines the rate at which the filter converges to its resultant state. Usually a faster convergence rate is a desired characteristic of an adaptive system. Convergence rate is not independent of all the other performance characteristics. There is usually a tradeoff, with convergence rate and other performance criteria .

B. Mean Square Error (MSE)

The MSE is a metric indicating how much a system can adapt to a given solution. A small MSE is an indication that the adaptive system has accurately modeled, predicted, adapted and/or converged to a solution for the system. There are a number of factors which will help to determine the MSE including, but not limited to; quantization noise, order of the adaptive system, measurement noise, and error of the gradient due to the finite step size.

Error = Desired signal – Enhanced signal
$e(n) = d(n) - y(n)$,
then MSE is

$$J(n) = \frac{1}{N}\sum_{i=0}^{N-1} e^2(n-i)$$

C. Computational Complexity

Computational complexity is particularly important in real time adaptive filter applications. When a real time system is being implemented, there are hardware limitations that may affect the performance of the system. A highly complex algorithm will require much greater hardware resources than a simplistic algorithm .

D. Log Spectral Distance

The log spectral distance (LSD), also referred to as log-spectral distortion, is a distance measure between two spectra. It calculates the average log-spectral distance between clean and noisy signals. The lower LSD, the higher the amount of noise cancellation.

The log spectra are much closer to the parameters used in a discriminator than power spectra. This spectral estimation performed better than spectral subtraction in noise immunity experiment. The log spectral distance between spectra P(w) and P(w) is defined as

$$LSD = \sqrt{\frac{1}{2\pi}\int_{-\pi}^{\pi}\left[10\log_{10}\frac{p(w)}{\bar{p}(w)}\right]^2 dw}$$

where P(w) is average power of noisy signal and $P\overline{(w)}$is average power of input signal.

E. Noise Reduction Ratio(NRR)

A representative part of the Noise Reduction Ratio (NRR) is defined as the ratio of average power of reference noise signal to error signal. It measures that how much noise is reduced from background. Note that NRR is calculated after the filter adaptation is finished (stationary condition).

$$NRR = 10\log_{10}\left[\frac{\sigma_x^2}{\sigma_e^2}\right]$$

$$\sigma_x^2 = \frac{1}{L}\sum_{n=0}^{L-1}[x(n)]^2, \sigma_e^2 = \frac{1}{L}\sum_{n=0}^{L-1}[e(n)]^2$$

CONCLUSION

A very efficient adaptive system based on IIR structures for noise suppression is proposed in this contribution. The main advantages of the present realization are:

- The adaptive system has a short time of adaptation about 100 iterations.
- The system is very simple and flexible, for comparison, here we adjust only 3 coefficients against 250-450 for conventional adaptive noise cancellation system.

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