

Applications of Adaptive Filtering

J. Gerardo Avalos, Juan C. Sanchez and Jose Velazquez
*National Polytechnic Institute
 Mexico*

1. Introduction

Owing to the powerful digital signal processors and the development of advanced adaptive algorithms there are a great number of different applications in which adaptive filters are used. The number of different applications in which adaptive techniques are being successfully used has increased enormously during the last two decades. There is a wide variety of configurations that could be applied in different fields such telecommunications, radar, sonar, video and audio signal processing, noise reduction, between others.

The efficiency of the adaptive filters mainly depends on the design technique used and the algorithm of adaptation. The adaptive filters can be analogical designs, digital or mixed which show their advantages and disadvantages, for example, the analogical filters are low power consuming and fast response, but they represent offset problems, which affect the operation of the adaptation algorithm (Shoval et al., 1995). The digital filters are offset free and offer an answer of greater precision. Also the adaptive filters can be a combination of different types of filters, like single-input or multi-input filters, linear or nonlinear, and finite impulse response FIR or infinite impulse response IIR filters.

The adaptation of the filter parameters is based on minimizing the mean squared error between the filter output and a desired signal. The most common adaptation algorithms are, Recursive Least Square (RLS), and the Least Mean Square (LMS), where RLS algorithm offers a higher convergence speed compared to the LMS algorithm, but as for computation complexity, the LMS algorithm maintains its advantage. Due to the computational simplicity, the LMS algorithm is most commonly used in the design and implementation of integrated adaptive filters. The LMS digital algorithm is based on the gradient search according to the equation (1).

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

Where $w(n)$ is the weights vector in the instant n , $w(n+1)$ is equal to the weights vector in $n+1$, $x(n)$ is the input signal sample vector which is stored in the filter delayed line, where $e(n)$ corresponds to the filter's error, which is the difference between the desired signal and the output filter's signal, and μ is the filter's convergence factor. The convergence factor μ determines the minimum square average error and the convergence speed. This factor is directly proportional to the convergence speed and indirectly proportional to the minimal error. Then a convergence speed and minimal error relation is established.

The application depends on the adaptive filter configuration used. The classical configurations of adaptive filtering are system identification, prediction, noise cancellation,

and inverse modeling. The differences between the configurations are given by the way the input, the desired and the output signals are used. The main objective of this chapter is to explain the typical configurations and it will focus on recent applications of adaptive filtering that are used in the real world.

2. System identification

The system identification is an approach to model an unknown system. In this configuration the unknown system is in parallel with an adaptive filter, and both are excited with the same signal. When the output MSE is minimized the filter represents the desired model.

The structure used for adaptive system identification is illustrated in figure 1, where $P(z)$ is an unknown system to be identified by an adaptive filter $W(z)$. The signal $x(n)$ excites $P(z)$ and $W(z)$, the desired signal $d(n)$ is the unknown system output, minimizing the difference of output signals $y(n)$ and $d(n)$, the characteristics of $P(z)$ can be determined.

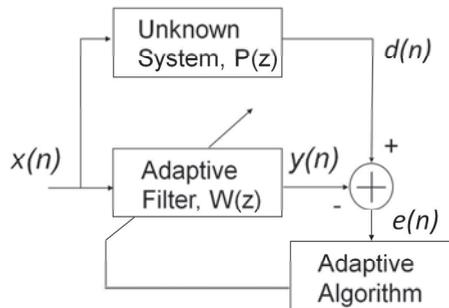


Fig. 1. Adaptive filter for system identification

The estimation error is given as (2)

$$e(n) = d(n) - y(n) = \sum_{l=0}^{L-1} [p(l) - w_1(n)] x(n-l) \quad (2)$$

Where $p(l)$ is the impulse response of the unknown plant, By choosing each $w_1(n)$ close to each $p(l)$, the error will be minimized. For using white noise as the excitation signal, minimizing $e(n)$ will force the $w_1(n)$ to approach $p(l)$, that is,

$$w_1(n) \approx p(l), l = 0, 1, \dots, L-1 \quad (3)$$

When the difference between the physical system response $d(n)$ and the adaptive model response $y(n)$ has been minimized, the adaptive model approximates $P(z)$ from the input/output viewpoint. When the plant is time varying, the adaptive algorithm has the task of keeping the modelling error small by continually tracking time variations of the plant dynamics.

Usually, the input signal is a wideband signal, in order to allow the adaptive filter to converge to a good model of the unknown system. If the input signal is a white noise, the best model for the unknown system is a system whose impulse response coincides with the $N + 1$ first samples of the unknown system impulse response. In the cases where the impulse response of the unknown system is of finite length and the adaptive filter is of sufficient order, the MSE becomes zero if there is no measurement noise (or channel noise).

In practical applications the measurement noise is unavoidable, and if it is uncorrelated with the input signal, the expected value of the adaptive-filter coefficients will coincide with the unknown-system impulse response samples. The output error will of course be the measurement noise (Diniz, 2008). Some real world applications of the system identification scheme include control systems and seismic exploration.

3. Linear predictor

The linear prediction estimates the values of a signal at a future time. This model is wide usually in speech processing applications such as speech coding in cellular telephony, speech enhancement, and speech recognition. In this configuration the desired signal is a forward version of the adaptive filter input signal. When the adaptive algorithm converges the filter represents a model for the input signal, this model can be used as a prediction model. The linear prediction system is shown in figure 2.

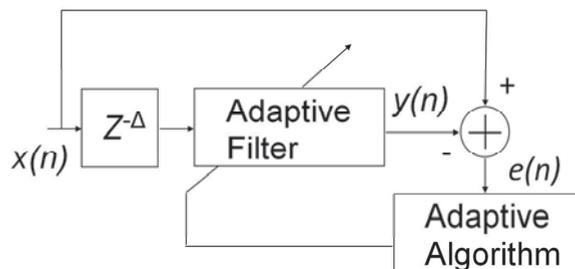


Fig. 2. Adaptive filter for linear prediction

The predictor output $y(n)$ is expressed as

$$y(n) = \sum_{l=0}^{L-1} w_1(n)x(n - \Delta - l) \quad (4)$$

Where Δ is the number of delay samples, so if we are using the LMS algorithm the coefficients are updated as

$$w(n + 1) = w(n) + \mu x(n - \Delta)e(n) \quad (5)$$

Where $x(n - \Delta) = [x(n - \Delta) \ x(n - \Delta - 1) \ \dots \ x(n - \Delta - L + 1)]^T$ is then delayed reference signal vector, and $e(n) = x(n) - y(n)$ is the prediction error. Proper selection of the prediction delay Δ allows improved frequency estimation performance for multiple sinusoids in white noise. A typical predictor's application is in linear prediction coding of speech signals, where the predictor's task is to estimate the speech parameters. These parameters are part of the coding information that is transmitted or stored along with other information inherent to the speech characteristics, such as pitch period, among others.

The adaptive signal predictor is also used for adaptive line enhancement (ALE), where the input signal is a narrowband signal (predictable) added to a wideband signal. After convergence, the predictor output will be an enhanced version of the narrowband signal. Yet another application of the signal predictor is the suppression of narrowband interference in a wideband signal. The input signal, in this case, has the same general characteristics of the ALE.

4. Inverse modeling

The inverse modeling is an application that can be used in the area of channel equalization, for example it is applied in modems to reduce channel distortion resulting from the high speed of data transmission over telephone channels. In order to compensate the channel distortion we need to use an equalizer, which is the inverse of the channel's transfer function.

High-speed data transmission through channels with severe distortion can be achieved in several ways, one way is to design the transmit and receive filters so that the combination of filters and channel results in an acceptable error from the combination of intersymbol interference and noise; and the other way is designing an equalizer in the receiver that counteracts the channel distortion. The second method is the most commonly used technology for data transmission applications.

Figure 3 shows an adaptive channel equalizer, the received signal $y(n)$ is different from the original signal $x(n)$ because it was distorted by the overall channel transfer function $C(z)$, which includes the transmit filter, the transmission medium, and the receive filter.

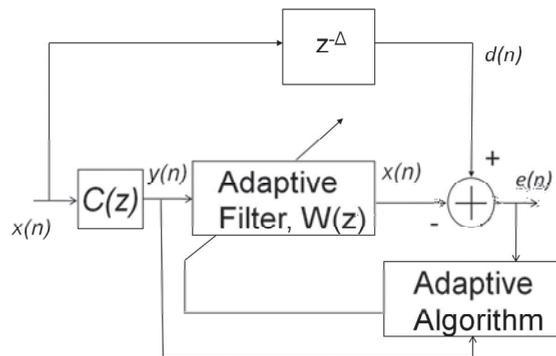


Fig. 3. Adaptive Channel equalizer

To recover the original signal $x(n)$, $y(n)$ must be processed using the equalizer $W(z)$, which is the inverse of the channel's transfer function $C(z)$ in order to compensate for the channel distortion. Therefore the equalizer must be designed by

$$W(z) = \frac{1}{C(z)} \quad (6)$$

In practice, the telephone channel is time varying and is unknown in the design stage due to variations in the transmission medium. Thus it is needed an adaptive equalizer that provides precise compensation over the time-varying channel. The adaptive filter requires the desired signal $d(n)$ for computing the error signal $e(n)$ for the LMS algorithm. An adaptive filter requires the desired signal $d(n)$ for computing the error signal $e(n)$ for the LMS algorithm.

The delayed version of the transmitted signal $x(n - \Delta)$ is the desired response for the adaptive equalizer $W(z)$. Since the adaptive filter is located in the receiver, the desired signal generated by the transmitter is not available at the receiver. The desired signal may be generated locally in the receiver using two methods. During the training stage, the adaptive equalizer coefficients are adjusted by transmitting a short training sequence. This known transmitted sequence is also generated in the receiver and is used as the desired signal $d(n)$ for the LMS algorithm.

After the short training period, the transmitter begins to transmit the data sequence. In the data mode, the output of the equalizer $x(n)$ is used by a decision device to produce binary data. Assuming that the output of the decision device is correct, the binary sequence can be used as the desired signal $d(n)$ to generate the error signal for the LMS algorithm.

5. Jammer suppression

Adaptive filtering can be a powerful tool for the rejection of narrowband interference in a direct sequence spread spectrum receiver. Figure 4 illustrates a jammer suppression system. In this case the output of the filter $y(n)$, is an estimate of the jammer, this signal is subtracted from the received signal $x(n)$, to yield an estimate of the spread spectrum.

To enhance the performance of the system a two-stage jammer suppressor is used. The adaptive line enhancer, which is essentially another adaptive filter, counteracts the effects of finite correlation which leads to partial cancellation of the desired signal. The number of coefficients required for either filter is moderate, but the sampling frequency may be well over 400 KHz.

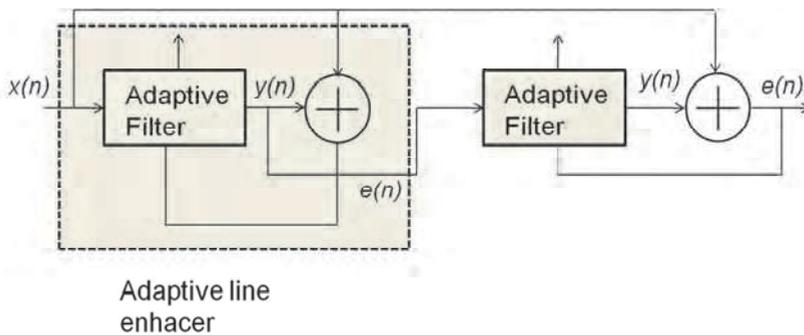


Fig. 4. Jammer suppression in direct sequence spread spectrum receiver

6. Adaptive notch filter

In certain situations, the primary input is a broadband signal corrupted by undesired narrowband (sinusoidal) interference. The conventional method of eliminating such sinusoidal interference is using a notch filter that is tuned to the frequency of the interference (Kuo et al., 2006). To design the filter, we need the precise frequency of the interference. The adaptive notch filter has the capability to track the frequency of the interference, and thus is especially useful when the interfering sinusoid drifts in frequency. A single-frequency adaptive notch filter with two adaptive weights is illustrated in figure 5, where the input signal is a cosine signal as

$$x(n) = x_0(n)A\cos(\omega_0 n) \quad (7)$$

A 90° phase shifter is used to produce the quadrature signal

$$x_1(n) = A\sin(\omega_0 n) \quad (8)$$

For a sinusoidal signal, two filter coefficients are needed. The reference input is used to estimate the composite sinusoidal interfering signal contained in the primary input $d(n)$.

The center frequency of the notch filter is equal to the frequency of the primary sinusoidal noise. Therefore, the noise at that frequency is attenuated. This adaptive notch filter provides a simple method for eliminating sinusoidal interference.

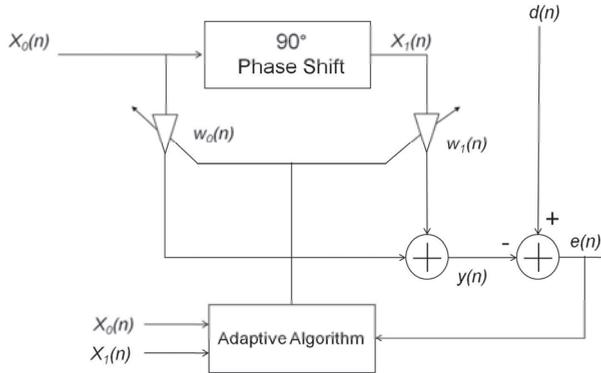


Fig. 5. Adaptive Notch Filter

7. Noise canceller

The noise cancellers are used to eliminate intense background noise. This configuration is applied in mobile phones and radio communications, because in some situations these devices are used in high-noise environments. Figure 6 shows an adaptive noise cancellation system.

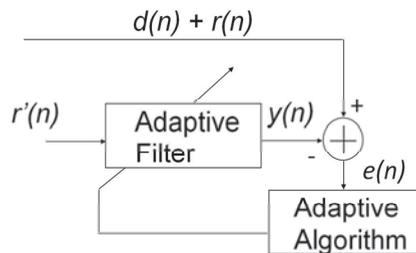


Fig. 6. Adaptive noise canceller system

The canceller employs a directional microphone to measure and estimate the instantaneous amplitude of ambient noise $r'(n)$, and another microphone is used to take the speech signal which is contaminated with noise $d(n) + r(n)$. The ambient noise is processed by the adaptive filter to make it equal to the noise contaminating the speech signal, and then is subtracted to cancel out the noise in the desired signal. In order to be effectively the ambient noise must be highly correlated with the noise components in the speech signal, if there is no access to the instantaneous value of the contaminating signal, the noise cannot be cancelled out, but it can be reduced using the statistics of the signal and the noise process. Figure 7 shows a voice signal with noise; those signals were used in noise canceller system implemented on a digital signal processor. The desired signal is a monaural audio signal with sampling frequency of 8 KHz. The noise signal is an undesired monaural musical piece with a sampling frequency of

11 KHz. As it can be seen in the image the desired signal is highly contaminated, so in this structure it must be used a fast adaptation algorithm in order to reach the convergence and eliminate all the unwanted components from the desired signal.

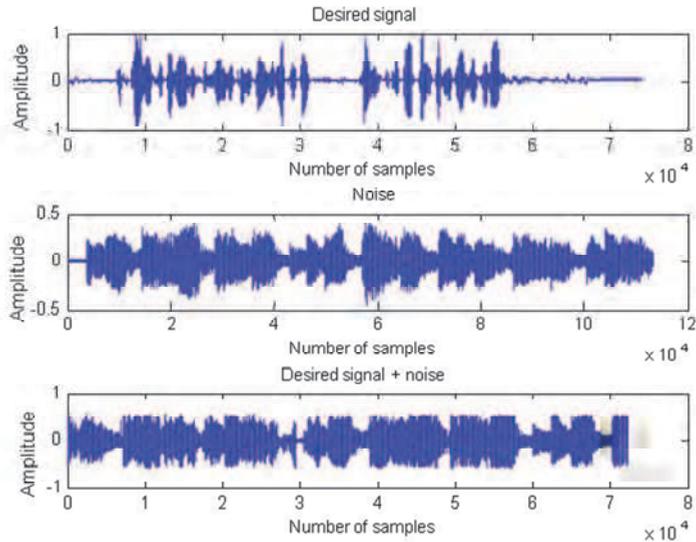


Fig. 7. Signals used in the noise canceller system

The frequency analysis of the signals used in the noise canceller system can be seen on the spectrograms of the figure 8. The figure shows that the output signal has some additional frequency components with respect to the input signal.

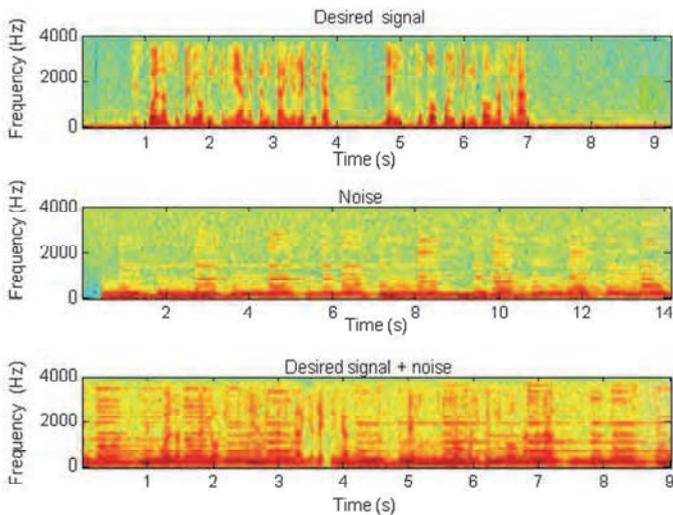


Fig. 8. Spectrograms of the signals used in the noise canceller system

The output of the noise canceller is the error signal, the figure 9 shows the error signal obtained when it is used an LMS algorithm. With the spectrogram of the signal it is shown that all the undesired frequency components were eliminated.

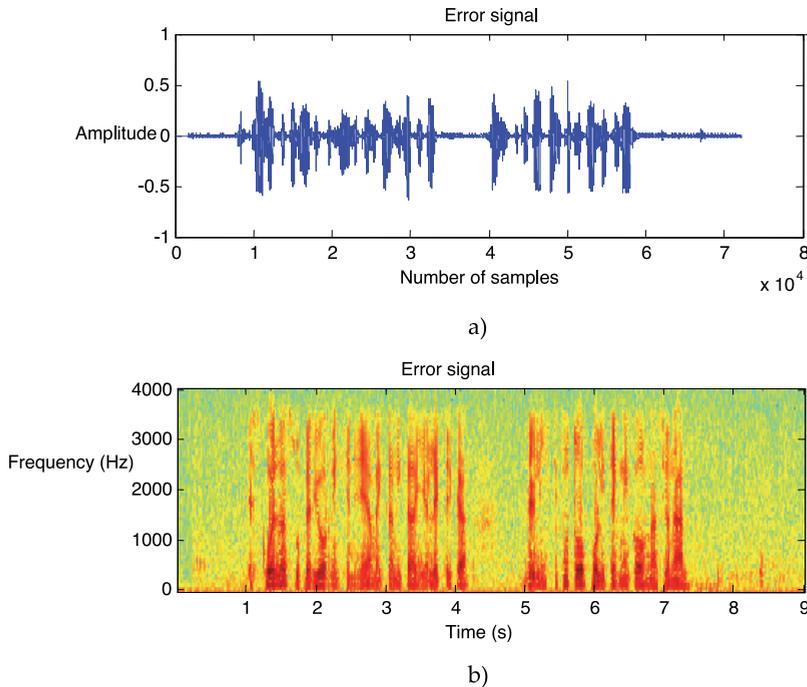


Fig. 9. a) Time waveform of the output signal b) Spectrogram of the output signal

The adaptive noise canceller system is used in many applications of active noise control (ANC), in aircrafts is used to cancel low-frequency noise inside vehicle cabins for passenger comfort. Most major aircraft manufacturers are developing such systems, mainly for noisy propeller-driven airplanes. In the automobile industry there are active noise cancellation systems designed to reduce road noise using microphones and speakers placed under the vehicle's seats.

Another application is active mufflers for engine exhaust pipes, which have been in use for a while on commercial compressors, generators, and such. With the price for ANC solutions dropping, even automotive manufacturers are now considering active mufflers as a replacement of the traditional baffled muffler for future production cars. The resultant reduction in engine back pressure is expected to result in a five to six percent decrease in fuel consumption for in-city driving.

Another application that has achieved widespread commercial success are active headphones to cancel low-frequency noise. The active headphones are equipped with microphones on outside of the ear cups that measure the noise arriving at the headphones. This noise is then being cancelled by sending the corresponding "anti-noise" to the headphones' speakers. For feedforward ANC, the unit also includes a microphone inside each ear cup to monitor the error - the part of the signal that has not been canceled by the

speakers in order to optimize the ANC algorithm. Very popular with pilots, active headphones are considered essential in noisy helicopters and propeller-powered airplanes.

7.1 Echo cancellation

In telecommunications, echo can severely affect the quality and intelligibility of voice conversation in telephone, teleconference or cabin communication systems. The perceived effect of an echo depends on its amplitude and time delay. In general, echoes with appreciable amplitudes and a delay of more than 1 ms can be noticeable. Echo cancellation is an important aspect of the design of modern telecommunications systems such as conventional wire-line telephones, hands-free phones, cellular mobile (wireless) phones, teleconference systems and in-car cabin communication systems.

In transmission networks the echoes are generated when a delayed and attenuated version of the signal sent by the local emitter to the distant receiver reaches the local receiver. These echo signals have their origin in the hybrid transformers which perform the two/four-wire conversion, in the impedance mismatches along the two-wire lines, and in some cases in acoustic couplings between loudspeakers and microphones in the subscriber sets.

The echo cancellation consists in modelling these unwanted couplings between local emitters and receivers and subtracting a synthetic echo from the real echo. According to the nature of the signals involved, the system will work as echo data canceller or voice echo canceller.

7.1.2 Voice echo canceller

Due to the characteristics of the speech signal, the voice echo cancellation system is somewhat different from the data echo canceller. The speech is a high level nonstationary signal, and due to the signal bandwidth and the velocity of the acoustic waves in the open air, the filters must have a very long number of coefficients. Also in order to reach a high level of performance and meet the expectations of the user, the voice echo canceller may have several other functions, like speech detection and denoising.

Figure 10 illustrates the operation of an adaptive line echo canceller. The speech signal on the line from speaker A to speaker B is input to the four/two-wire hybrid B and to the echo canceller. The echo canceller monitors the signal on line from B to A and attempts to model the echo path and synthesise a replica of the echo of speaker A. This replica is used to subtract and cancel out the echo of speaker A on the line from B to A. The echo canceller is basically an adaptive linear filter. The coefficients of the filter are adapted so that the energy of the signal on the line is minimised.

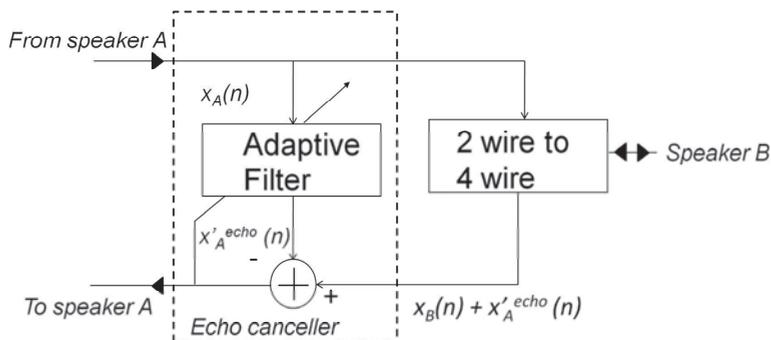


Fig. 10. Adaptive echo cancellation system

Assuming that the signal of the line from speaker B to speaker A, $y_B(n)$, is composed of the speech of speaker B, $x_B(n)$, plus the echo of speaker A, $x_A^{echo}(n)$,

$$y_B(n) = x_B(n) + x_A^{echo}(n) \quad (9)$$

Speech and echo signals are not simultaneously present on a phone line unless both speakers are speaking simultaneously. Assuming that the truncated impulse response of the echo path is modelled by an FIR filter, the output estimate of the synthesised echo signal can be expressed as

$$x_A'^{echo}(n) = \sum_{l=0}^P h_l(n) x_A(n-l) \quad (10)$$

Where $h_l(n)$ are the time varying coefficients of an adaptive FIR filter model of the echo path and $x_A'^{echo}(n)$ is an estimate of the echo of speaker A on the line from speaker B to speaker A. The residual echo signal, or the error signal, after echo subtraction is given by

$$e(n) = y_B(n) - x_A'^{echo}(n) = x_B(n) + x_A^{echo}(n) - \sum_{l=0}^P h_l(n) x_A(n-l) \quad (11)$$

For those time instants when speaker A is talking and speaker B is listening and silent, and only echo is present from line B to A, we have

$$e(m) = x_A'^{echo}(m) = x_A^{echo}(m) - x_A'^{echo}(m) = x_A^{echo}(m) - \sum_{l=0}^P h_l(m) x_A(m-l) \quad (12)$$

Where $x_A'^{echo}(n)$ is the residual echo.

In some cases it may happen the double talk situation, in this case both users talk at the same time, and simultaneous bidirectional transmission takes place. In this way it could be produced misalignment of the coefficients and a drop in echo attenuation, one way to solve this problem is holding the coefficients during double talk, but for this it is needed a double-talk detector. The performance of double-talk detectors is crucial for the comfort of the users.

7.1.3 Data echo canceller

Echo cancellation becomes more complex with the increasing integration of wireline telephone systems and mobile cellular systems, and the use of digital transmission methods such as asynchronous transfer mode (ATM) for integrated transmission of data, image and voice.

Those systems use full-duplex transmission data signals that are transmitted simultaneously in two directions and in the same frequency bands, meanwhile in half-duplex transmission just one direction are used at a time. The figure 11 shows the principle of full-duplex transmission. The signal $x_A(N)$ is sent from terminal A to terminal B through a two wire line. The signal $y(n)$ at the input of the receiver of terminal A consists of two components, a signal from the terminal B ($y_B(n)$), which is the useful data signal, and the returned unwanted echo generated from $x_A(n)$. $H(z)$ is a filter that is going to generate a synthetic echo $y'(n)$ as close as possible to $x_A(n)$, after subtraction, the output error $e(n)$ is kept sufficiently close to $y_B(n)$ to make the transmission of data from terminal B to terminal A satisfactory.

The number of coefficients (N) of the adaptive filter is derived from the duration of the echo impulse response that has to be compensated, taking into account the sampling frequency. In order to calculate the number of coefficients we could use

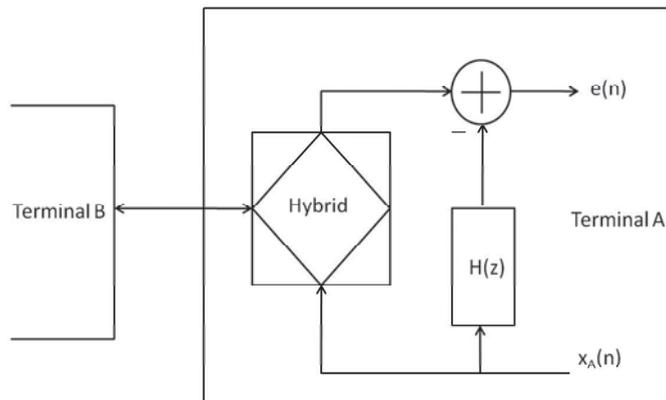


Fig. 11. Echo cancellation for full-duplex transmission

$$N = (2D/v) f_s \quad (1)$$

Where N is the number of coefficients, D is the length of the line, v is the electrical signal velocity over the subscriber line and f_s is the sampling frequency (Bellanger, 2001). Since the characteristics of the transmission line may change with time it is necessary to implement an adaptive filter.

7.1.4 Acoustic echo

Acoustic echo results from a feedback path set up between the speaker and the microphone in a mobile phone, hands-free phone, teleconference or hearing aid system. Acoustic echo is reflected from a multitude of different surfaces, such as walls, ceilings and floors, and travels through different paths. If the time delay is not too long, then the acoustic echo may be perceived as a soft reverberation, and may add to the artistic quality of the sound; concert halls and church halls with desirable reverberation characteristics can enhance the quality of a musical performance.

Acoustic echo can result from a combination of direct acoustic coupling and multipath effect where the sound wave is reflected from various surfaces and then picked up by the microphone. In its worst case, acoustic feedback can result in howling if a significant proportion of the sound energy transmitted by the loudspeaker is received back at the microphone and circulated in the feedback loop.

The most effective method of acoustic feedback removal is the use of an adaptive feedback cancellation system (AFC). Fig. 12 illustrates a model of an acoustic feedback environment, comprising a microphone, a loudspeaker and the reverberating space of a room (Vaseghi, 2006). The z transfer function of a linear model of the acoustic feedback environment may be expressed as

$$H(z) = \frac{G(z)}{1 - G(z)A(z)} \quad (13)$$

Where $G(z)$ is the z transfer function model for the microphone loudspeaker system and $A(z)$ is the z transfer function model of reverberations and multipath reflections of a room environment. Assuming that the microphone loudspeaker combination has a flat frequency response with a gain G , the equation can be simplified to

$$H(z) = \frac{G}{1-GA(z)} \quad (14)$$

Owing to the reverberation character of the room, the acoustic feedback path $A(z)$ is itself a feedback system. The reverberating characteristics of the acoustic environment may be modelled by an all-pole linear predictive model, or alternatively a relatively long FIR model. The equivalent time-domain input/output relation for the linear filter model of equation (4) is given by the following difference equation

$$y(n) = \sum_{l=0}^P a_l(n)y(n-l) + Gx(n) \quad (15)$$

Where $a_l(n)$ is the coefficient of an all pole linear feedback model of the reverberating room environment, G is the microphone loudspeaker amplitude gain factor, and $x(n)$ and $y(n)$ are the time domain input and output signals of the microphone loudspeaker system.

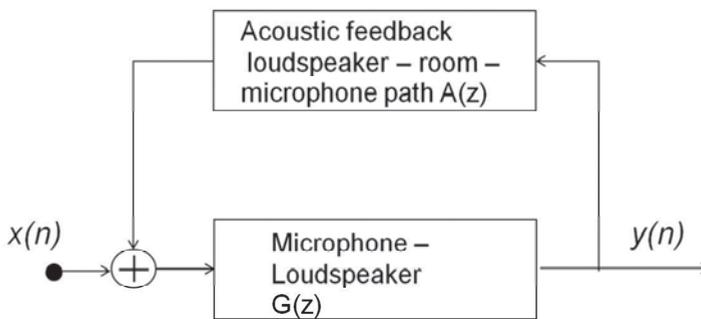


Fig. 12. Acoustic feedback model

The most successful acoustic feedback control systems are based on adaptive estimation and cancellation of the feedback signal. As in a line echo canceller, an adaptive acoustic feedback canceller attempts to synthesise a replica of the acoustic feedback. The problem of acoustic echo cancellation is more complex than line echo cancellation for a number of reasons. First, acoustic echo is usually much longer (up to a second) than terrestrial telephone line echoes. In fact, the delay of an acoustic echo is similar to or more than a line echo routed via a geostationary satellite system. The large delay of an acoustic echo path implies that impractically large filters on the order of a few thousand coefficients may be required. An important application of acoustic feedback cancellation is in hearing aid systems.

7.1.5 Multiple-input multiple-output (MIMO) echo cancellation

Multiple-input multiple-output (MIMO) echo-cancellation systems have applications in car cabin communications systems, stereophonic teleconferencing systems and conference halls. Stereophonic echo cancellation systems have been developed relatively recently and MIMO systems are still the subject of ongoing research and development. In a typical MIMO system there are P speakers and Q microphones in the room. As there is an acoustic feedback path set up between each speaker and each microphone, there are altogether $P \times Q$ such acoustic feedback paths that need to be modelled and estimated. The truncated impulse response of each acoustic path from loudspeaker i to microphone j is modelled by an FIR filter h_{ij} . The truncated impulse response of each acoustic path from a human speaker

i to microphone j is modelled by an FIR filter, g_{ij} . For a large number of speakers and microphones, the modelling and identification of the numerous acoustic channels becomes a major problem due to the correlations of the echo signals, from a common number of sources, propagating through different channels, as discussed below.

7.2 Adaptive feedback cancellation in hearing aids

The hearing-aid processing amplifies the input signal to compensate for the hearing loss of the users. When this amplification is larger than the attenuation of the feedback path, instability occurs and usually results in feedback whistling, which limits the maximum gain that can be achieved.

Acoustic feedback in hearing aids refers to the acoustical coupling between the loudspeaker (also known as the receiver) and the microphone of the hearing aid. Because of this coupling, the hearing aid produces a severe distortion of the desired signal and an annoying howling sound when the gain is increased.

If the Feedback transfer function was known, it can be compensated for in the hardware, but the problem here is the time variability of the dynamics, caused by a change in interference characteristics. Some possible causes of this problem are hugs or objects like a telephone coming close to the ear.

There are several techniques to reduce the negative effects introduced by acoustic feedback. They can be broadly classified into feedforward suppression and feedback cancellation techniques. In feedforward suppression techniques, the regular signal processing path of the hearing aid is modified in such a way that it is stable in conjunction with the feedback path. The most common technique is the use of a notch filter. In a notch filter, the gain is reduced in a narrow frequency band around the critical frequencies whenever feedback occurs. Nevertheless feedforward suppression techniques all compromise the basic frequency response of the hearing aid, and, hence, may seriously affect the sound quality (Spriet et al., 2006). A more promising solution for acoustic feedback is the use of a feedback cancellation system.

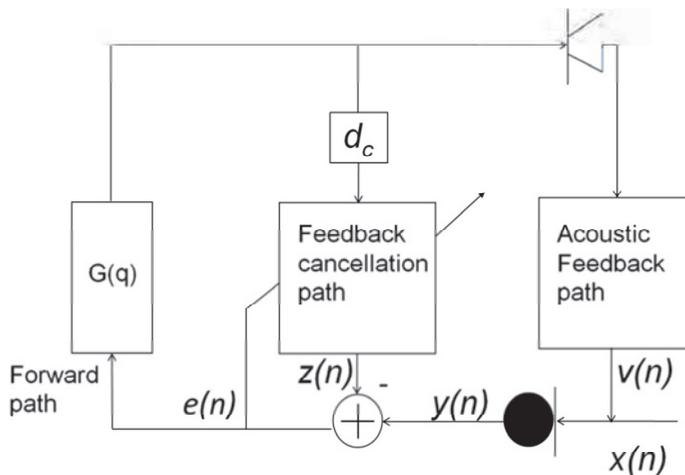


Fig. 13. Adaptive feedback canceller

Figure 13 illustrates an adaptive feedback canceller, which produces an estimate $z(n)$ of the feedback signal $v(n)$ and subtracts this estimate $z(n)$ from the microphone signal, so that, ideally, only the desired signal is preserved at the input of the forward path. Since the acoustic path between the loudspeaker and the microphone can vary significantly depending on the acoustical environment, the feedback canceller must be adaptive.

When the external input signal is correlated with the receiver input signal, the estimate of the feedback path is biased. This so-called "bias problem" results in a large modeling error and a cancellation of the desired signal (Ma, 2010).

7.3 Foetal monitoring, cancelling of maternal ECG during labour

Information derived from the foetal electrocardiogram (ECG), such as the foetal heart rate pattern, is valuable in assessing the condition of the baby before or during the childbirth. The ECG derived from electrodes placed on the mother's abdomen is susceptible to contamination from much larger background noise (for example muscle activity and foetal motion) and the mother's own ECG.

Considering the problem as an adaptive noise cancellation, where foetal ECG is a desired signal $d(n)$, corrupted by the maternal signal $r(n)$, a kind of additive noise. The measured foetal signal (MFECG(n)) from foetal lead can be expressed as

$$\text{MFECG}(n) = d(n) + r(n) \quad (16)$$

Another measurement MMECG(n) from maternal lead is given as a reference signal, that is correlated with $r(n)$ and uncorrelated with $d(n)$. MMECG can be used to estimate the noise $r(n)$ by minimizing the mean square error. Figure 14 shows the block diagram for the enhancement of foetal ECG.

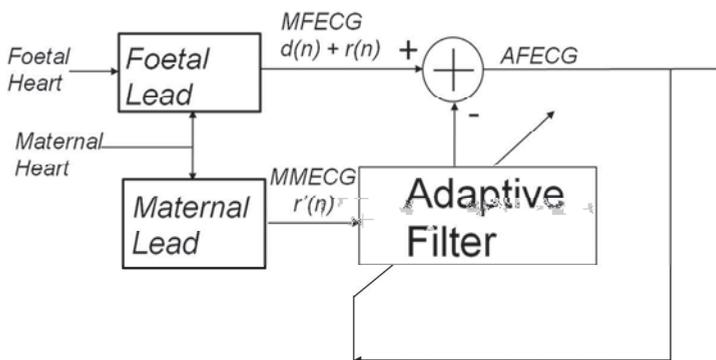


Fig. 14. Adaptive cancelling of maternal ECG in foetal ECG

An adaptive filter is used to estimate maternal components in measured foetal ECG (MFECG) from measured maternal ECG (MMECG). The estimated components then are subtracted from the MFECG to obtain adaptive filtered foetal ECG (AFECG), in which maternal components are suppressed. Other artefacts, such as muscular contraction from maternal body and foetal movement, will induce baseline drift in the MFECG (Chen et al., 2000).

7.4 Removal of ocular artifacts from electro-encephalogram by adaptive filtering

The eye forms an electric dipole, where the cornea is positive and the retina is negative. When the eye moves (saccade, blink or other movements), the electric field around the eye changes, producing an electrical signal known as the electro-oculogram (EOG). As this signal propagates over the scalp, it appears in the recorded electro-encephalogram (EEG) as noise or artifacts that present serious problems in EEG interpretation and analysis. There are at least two kinds of EOG artifact to be removed: those produced by the vertical eye movement (the corresponding EOG is called VEOG) and those produced by the horizontal eye movement (HEOG). Consequently, a noise canceller with two reference inputs is used in this application (He et al., 2004).

Fig. 15 shows the EOG noise canceller. The primary input to the system is the EEG signal $s(n)$, picked up by a particular electrode. This signal is modelled as a mixture of a true EEG $x(n)$ and a noise component $r(n)$. $v(n)$ and $v'(n)$ are the two reference inputs, VEOG and HEOG, respectively. $v(n)$ and $v'(n)$ are correlated, in some unknown way, with the noise component $r(n)$ in the primary input. The desired output from the noise canceller $e(n)$ is the corrected, or clean, EEG.

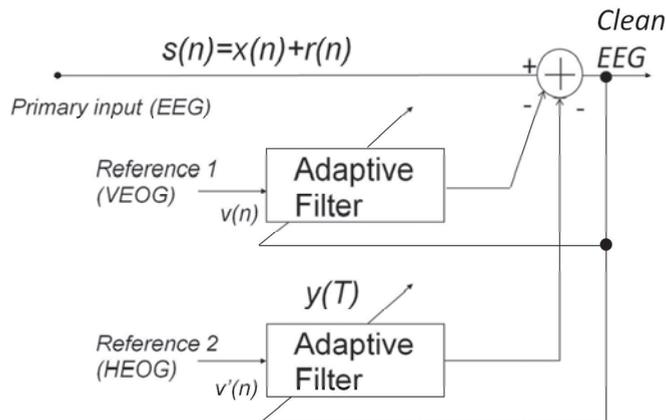


Fig. 15. EOG noise canceller

7.5 Application of adaptive noise cancelling filters in AC electrical measurements

Through adaptive noise cancellation it could be improved the ac electrical measurements. Often ac measurement circuits are influenced by noise caused by line frequency beat. The figure 16 shows a system that cancels the line frequency beat. An ADC is used to sample the suitably divided down line voltage in order to determine the phase relative to the signal channel, which is sampled with a second ADC. The phase data is used as the noise input to an adaptive noise-cancelling filter used to cancel the effect on the transconductance amplifier output data (Wright et al., 2010).

Another common interference in ac measurement circuits is the coupling of the magnetic field generated by a nearby source. In such situations it may be possible to use an adaptive interference cancelling system with a simple coil system to measure the ambient magnetic field that causes the unwanted interference and then remove this interference from data obtained from a measurement circuit.

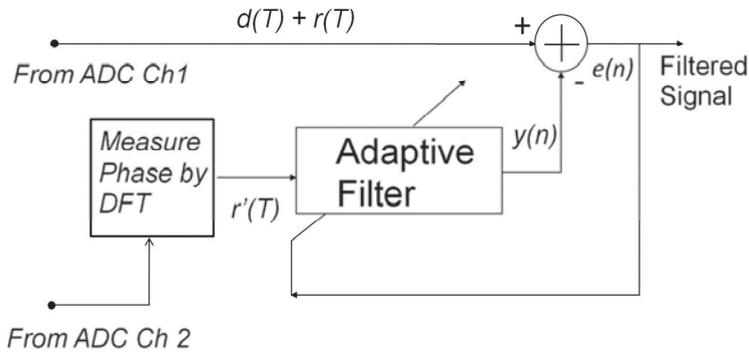


Fig. 16. Line beat Adaptive canceller.

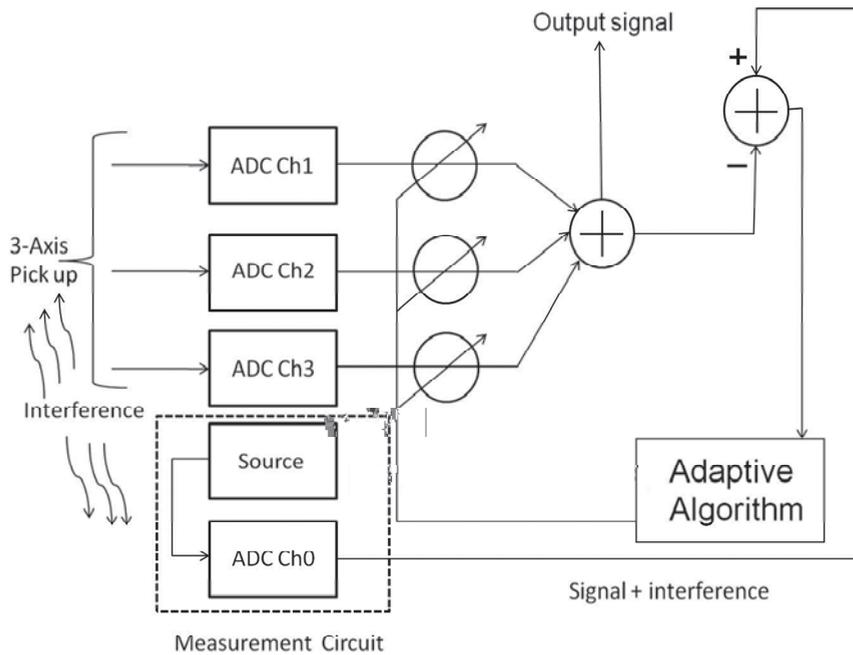


Fig. 17. Three Axis linear combiner for interference cancellation

Figure 17 shows a 3-axis magnetic field sensor which is connected to a separate analogue to digital converter (ADC). A fourth ADC is used to sample the “signal” simultaneously with the 3-axis data. The three “noise” channel ADCs are the inputs to the three channels of a three-way linear combiner (Wright et al., 2010).

8. Conclusion

In recent years, the development and commercial availability of increasingly powerful and affordable digital computers has been accompanied by the development of advanced digital signal processing algorithms for a wide variety of applications; therefore the use of adaptive filters is bigger every day.

Adaptive filters are used for estimation of nonstationary signals and systems, or in applications where a sample-by sample adaptation of a process and/or a low processing delay is required.

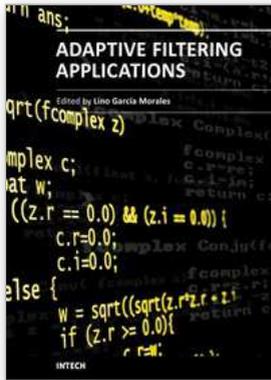
In this chapter, we described some of the most used adaptive filtering applications. The material presented here forms the basis to understand the behavior of most adaptive-filtering structures in practical implementations. The main objective was to illustrate how the adaptive-filtering is applied to solve practical problems.

The distinctive feature of each application is the way the adaptive filter input signal and the desired signal are chosen. Once these signals are determined, any known properties of them can be used to understand the expected behavior of the adaptive filter when attempting to minimize the chosen objective function. The efficiency of the adaptive filters mainly depends on the used technique of design and the algorithm of adaptation.

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Adaptive Filtering Applications

Edited by Dr Lino Garcia

ISBN 978-953-307-306-4

Hard cover, 400 pages

Publisher InTech

Published online 24, June, 2011

Published in print edition June, 2011

Adaptive filtering is useful in any application where the signals or the modeled system vary over time. The configuration of the system and, in particular, the position where the adaptive processor is placed generate different areas or application fields such as: prediction, system identification and modeling, equalization, cancellation of interference, etc. which are very important in many disciplines such as control systems, communications, signal processing, acoustics, voice, sound and image, etc. The book consists of noise and echo cancellation, medical applications, communications systems and others hardly joined by their heterogeneity. Each application is a case study with rigor that shows weakness/strength of the method used, assesses its suitability and suggests new forms and areas of use. The problems are becoming increasingly complex and applications must be adapted to solve them. The adaptive filters have proven to be useful in these environments of multiple input/output, variant-time behaviors, and long and complex transfer functions effectively, but fundamentally they still have to evolve. This book is a demonstration of this and a small illustration of everything that is to come.

How to reference

In order to correctly reference this scholarly work, feel free to copy and paste the following:

J. Gerardo Avalos, Juan C. Sanchez and Jose Velazquez (2011). Applications of Adaptive Filtering, Adaptive Filtering Applications, Dr Lino Garcia (Ed.), ISBN: 978-953-307-306-4, InTech, Available from: <http://www.intechopen.com/books/adaptive-filtering-applications/applications-of-adaptive-filtering>

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InTech Europe

University Campus STeP Ri
Slavka Krautzeka 83/A
51000 Rijeka, Croatia
Phone: +385 (51) 770 447
Fax: +385 (51) 686 166
www.intechopen.com

InTech China

Unit 405, Office Block, Hotel Equatorial Shanghai
No.65, Yan An Road (West), Shanghai, 200040, China
中国上海市延安西路65号上海国际贵都大饭店办公楼405单元
Phone: +86-21-62489820
Fax: +86-21-62489821

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