Volume XXIII 2020
ISSUE no.1
MBNA Publishing House Constanta 2020

Scientific Bulletin of Naval Academy

SBNA PAPER • OPEN ACCESS

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To cite this article: Travediu Ana-Maria, Manoliu Mitica-Valentin and Racuciu Ciprian, Scientific Bulletin of Naval Academy, Vol. XXIII 2020, pg.90-96.

Available online at www.anmb.ro

ISSN: 2392-8956; ISSN-L: 1454-864X

doi: 10.21279/1454-864X-20-I1-012

Improve channel signal quality using adaptive filters

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Abstract. A channel can be modelled by trying to calculate the physical processes which modify the transmitted signal. In this article we simulated the adaptive equalization of some communication channels. A system of data transmission in the basic band will be considered in order to illustrate the principle of adaptive equalization. In such a system, the data sequence is applied, upon emission, to a formation filter(low pass filter), with the purpose of limiting the occupied bandwidth. A relatively wide band noise is also introduced on the communication channel. The transmission channel can also be modelled as a linear filter, this is represented by the distortion filter. We used another filter to avoid the distortion introduced by the channel, with the role of equalizer. This should compensate for the distortion introduced by the channel. The channel transfer function is not generally known, and in addition, it may vary slowly over time. Therefore, the need for this filter appears to be adaptive. In order to perform the adaptation initially, a training sequence, known at the reception, can be transmitted first and the output of the filter is compared with a locally generated reference signal, but identical to the one emitted. After this initial adaptation phase, the transmitter transmits the information symbols, the useful signal, thus the application can be used to improve the signal quality of maritime communications channels.

1. Introduction

Radio was originally a method of transmitting sounds through radio waves, which by their nature are electromagnetic waves. Today a wide range of different signals are transmitted over the radio, including images, television and huge data streams. Radio waves travel through the air and can pass through most non-metallic bodies including the human body.[1]

It is unanimously admitted that the first who was able to make a broadcast and reception from radio was the German physicist Heinrich Hertz in 1887, which was based on his own theoretical physics studies, to which were added those of his predecessors in especially Maxwell. Another pioneer who contributed to the development of the radio, was the Russian physicist Aleksander Stepanovici Popov, with his storm recorder, 1895, who made the first systematic receptions, being the one to whom the invention of the antenna is attributed. Guglielmo Marconi, systematizing the data up to him, offered the world, in 1896, the first practical emission and reception system based on electromagnetic waves, based on the Tesla apparatus. The existence of radio waves was made known to the general public in particular by Guglielmo Marconi, an Italian inventor operating in England. Croatian physicist in America Nikola Tesla has contributed, along with several other inventors, to the creation of the first radio. He built a system that could transmit and receive radio signals from a distance of almost 3 km.

In 1895 he sent a radio signal for the first time; In 1907 he first received a radio signal from Canada, namely the "x" sign in the Morse Code.

With this began to develop wireless telegraphy and the use of Morse code, which were very important especially for communication between ships in the event of disasters at sea. The first to send a voice message through radio waves was Reginald Fessenden in 1900. Nikola Tesla started building the first radio station in 1900, but for lack of funds abandoned the idea. However, he is considered the inventor of the idea of radio stations with programs.

2. Pre-processing

The voice signal was recorded in a wave file using the audacity program. The recording is mono (single channel), the sampling frequency is 48 kHz, the number of quantization bits is 16 and it was obtained on export, total duration of the recorded signal is 8-10 seconds.[2]

We eliminated the continuous component of the signal. This implementation allowed us to create a function that receives as input the original signal and returns the signal without the continuous component. The resulting signal is denoted by s1. The continuous component is also the average of the signal and has the value of 1.4998e-006. We downsampled the signal with a factor corresponding to the reduction of the sampling frequency at 16 kHz. The sampling frequency can be reduced by M times decimating the signal x [n]. The signal decimated by the factor M is:

$$y[n] \equiv x[n] \downarrow M = x[Mn].$$

The old sampling frequency is 48 kHz. The new sampling frequency is 16 kHz.

$$fe' = fe / M \Rightarrow M = fe / fe'.$$

The decimation factor is 3, integer. If the spectrum of the initial signal extends beyond the frequency π/M, then the spectrum of the decimated signal does not generally have the same shape as the spectrum of the initial signal. The aliasing phenomenon appears. To avoid aliasing (but affecting the initial spectrum), a low-pass filter is used that cuts the frequencies above π/M.

By default, the decimated function filters the input data with a Cebîevev type I filter, of the 8th order, pass-down, before re-sampling, to avoid spectral aliasing. Filtering the signal through an Low Pass Filter with the cutoff frequency of 3.4 kHz.

We used a low-pass filter, Butterworth, of order 20, because the higher the order of the filter, the more the characteristic of the frequency response of the filter is closer to the ideal one. The voice signal has the band between 20Hz-4KHz (narrowband signal). The cut-off frequency is 3.4KHz so that it is not at the limit, an ideal filter cannot be implemented in reality. Most of the voice signal (80-90%) of it has a bandwidth of 3.4KHz.

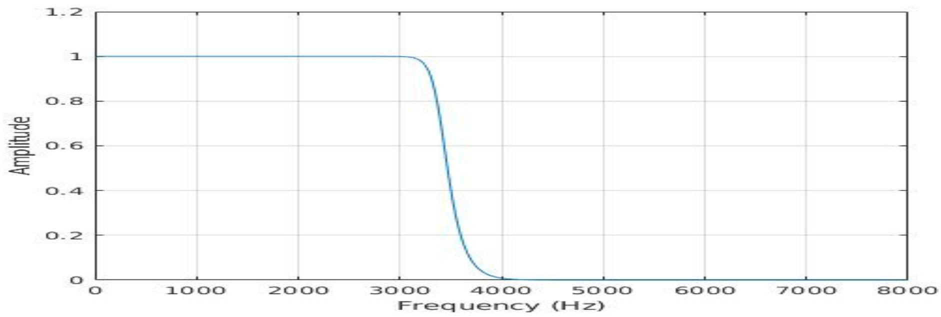

Figure 1. Digital distortioning filter

In Figure 1 is represented the implementation in MATLAB of the digital distortion filter, starting from the coefficients b and a. The distorted signal is obtained at the output of this system, following the application of the useful signal at its input. The distorted signal is denoted by x.

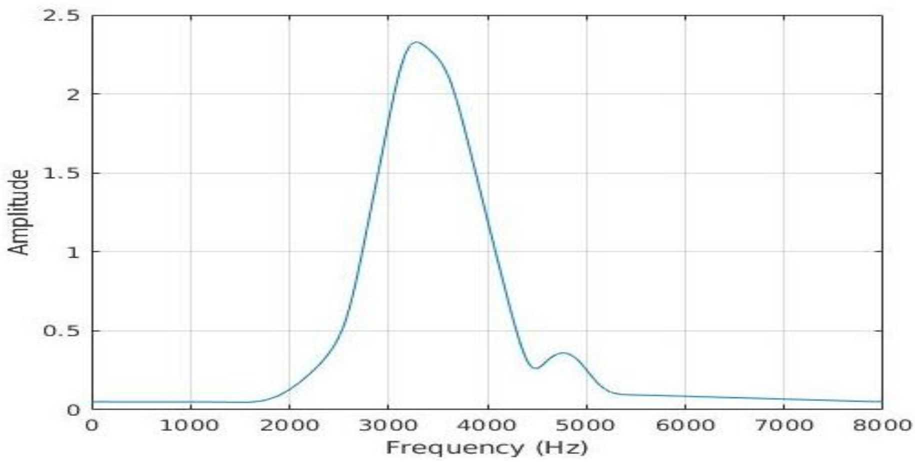

Figure 2. The spectrum of the distortion filter

The filter presented in Figure 2 attenuates the low frequencies and amplifies the high frequencies.

In Figure 3 we investigated the signal before and after the filter process (in the frequency and time domain) and the frequency response of the distortion filter (unknown system), to determine the degree of distortion between the signals in Figure 4. We observed in Figure 5 that the signal, the spectrum and the PDS (power spectral density), estimated using Welch method, are affected by this pre-processing technique.[4]

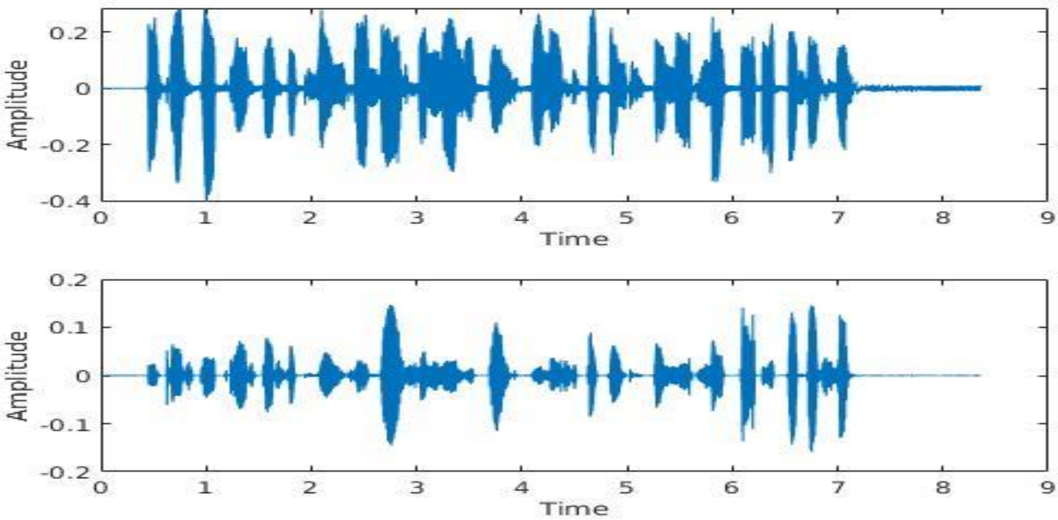

Figure 3. Comparison between the signal before and after preprocessing

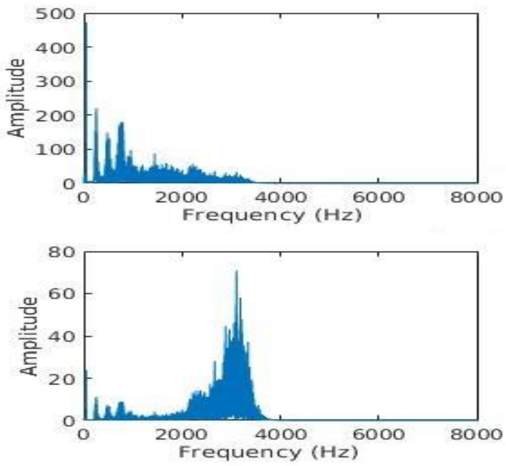

Figure 4 Comparison between the specter of the signal before and after preprocessing [12]

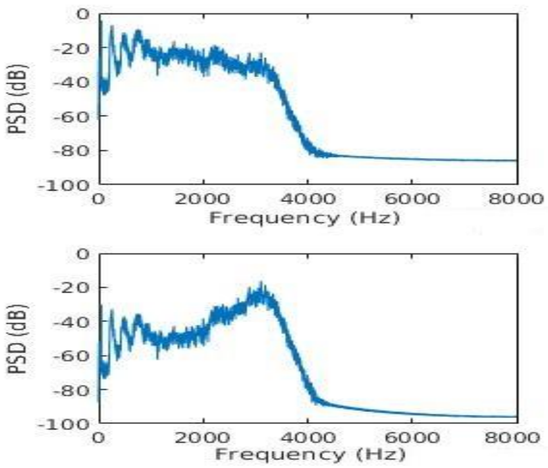

Figure 5. Comparision between Power Spectral Density of the signal before and after preprocessing [13]

3. Implementationof the nLMS algorithm in the "inverse modeling" configuration

In Figure 6 implementated in MATLAB an adaptive filter with the nLMS algorithm in the "inverse modeling" configuration.

An adaptive filter is a filter capable of changing its parameters, in order to optimize its characteristics, based on a recursive algorithm. This algorithm adjusts the iterative parameters, minimizing a certain criterion by lowering the maximum slope on its performance surface in order to reach the minimum.[3]

Reverse modeling: this configuration allows to identify an unknown system, by connecting it in cascade with the adaptive filter. In the event that the error is null, it would mean that the global transfer function of the unknown system and the adaptive filter would be reduced to a delay. So generally, the adaptive filter transfer function will approximate the inverse of the unknown system transfer function, an identity not being possible due to the inevitable presence of noise. At the same time, this scheme has the property to eliminate the effect of an unwanted transfer function.

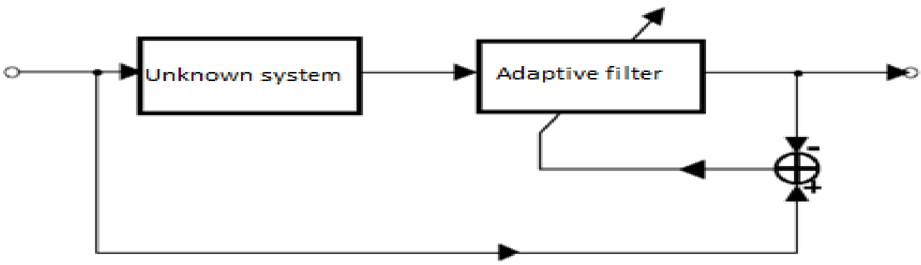

Figure 6. The "inverse modeling" configuration

The error is used by the adaptive algorithm for the recursive modification of the coefficients. In order to assess the optimal character of the filter, a "cost function" must be defined. In our case, the weighted sum of the error squares. The adaptive filter is trapped using an auxiliary test signal as the desired signal. To initially adapt, a test sequence, known at reception, can be transmitted first and the output of the filter is compared with a locally generated reference signal, but identical to the one emitted. After this initial adaptation phase, which usually lasts less than one second, the transmitter emites information symbols.

The algorithm is convergent on average if the mean value of the impulse response of the adaptive filter, tends toward the optimal vector when the number of samples tends towards infinity.

Adaptation is made with a white noise because this signal has constant power over its entire spectrum and respects the so-called independence assumptions.[7]

In the frequency, the effects of the distortion filter on the initial signal are very well observed: the low frequencies (about 2 kHz) are greatly attenuated and the high frequencies (3 to 4 kHz) are amplified.

In the analysis of the test sequence in time, is observed the adaptation time of the filter.

An important role in adaptation has the learning parameter μ. It represents the rate of adaptation of the filter. A high value for μ means a faster convergence, but a higher final mismatch, while a small value for μ means a slower adaptation, but a smaller final mismatch, but the optimal values for the impulse response are only achieved theoretically.

The NLMS algorithm uses mu depending on the power of the input signal which varies if the signal is stationary and the filter order. So the signal strength affects the convergence speed, even the filter convergence.[11]

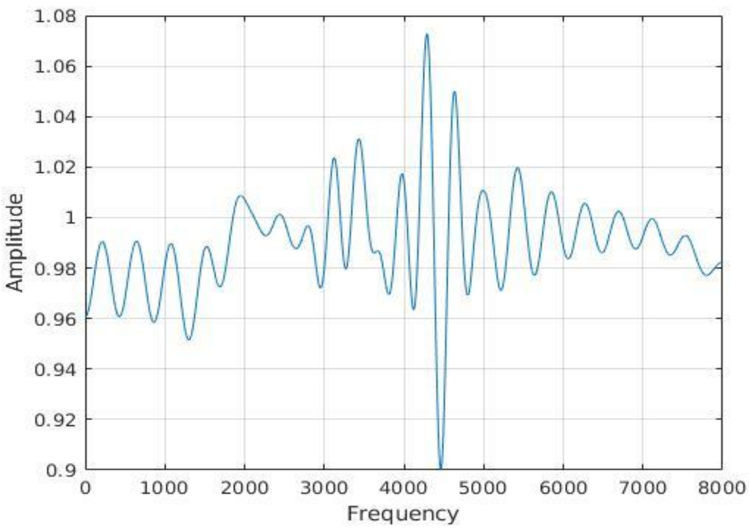

Figure 7. The frequency product of the filters

Ideally, in Figure 7 the frequency product of the filters should be equal to 1 for the error to be zero as shown in the figure above. It can be seen that the adaptation is not immediate, it takes a while to adapt the filter, but after the adaptation the product of the two filters tends to 1, so the error tends to zero[5].

Considering that from this moment, the coefficients of the filter found after the initial training operation are fixed, this filter will be used to filter the distorted signal x and to obtain at the output the recovered useful signal, denoted by y; Under these conditions, the y signal will be the best possible approximation for the s signal.

It is observed that if we have access to the undistorted signal as the desired signal, the filtering is better, but the filter coefficients also change so that the filter is convergent.

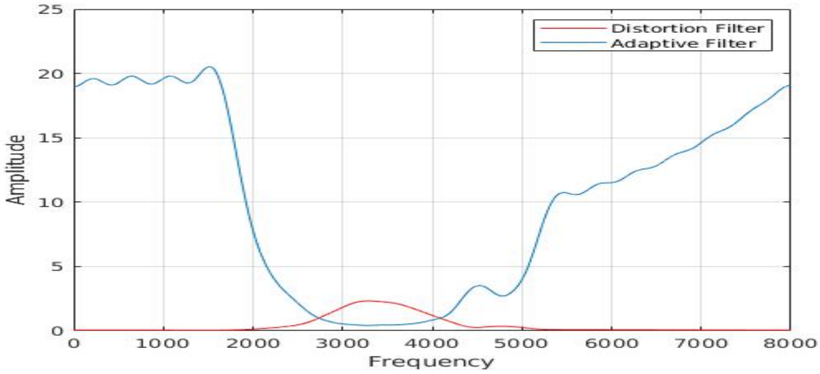

Figure 8. The frequency comparison between the dirtortion filter and the adaptive filter

In Figure 8 we using another method by which we can solve this problem is to use an adaptive filter in the "system identification" configuration. The adaptive filter will try to mimic the unknown system. So I will first identify the distortion filter and then pass the signal through a filter that has the impulse response even the inverse of the distortion filter impulse response.

Because the adaptive filter is FIR, the new filter will be IIR, only with poles (the coefficients of the new filter are the coefficients b of the adaptive filter).[6]

4. Implementationof the nLMS algorithm in the "inverse modeling" configuration

We implementated in MATLAB an adaptive filter with the nLMS algorithm in the " system identification " configuration.[10]

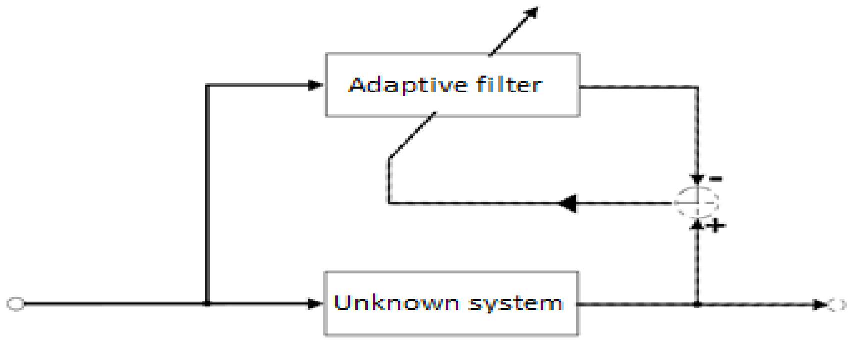

Figure 9. The "inverse modeling" configuration

We observed in Figure 9 that both in frequency and after power analysis that the adaptive filter mimics the behavior of the "unknown system" (the adaptive filter)

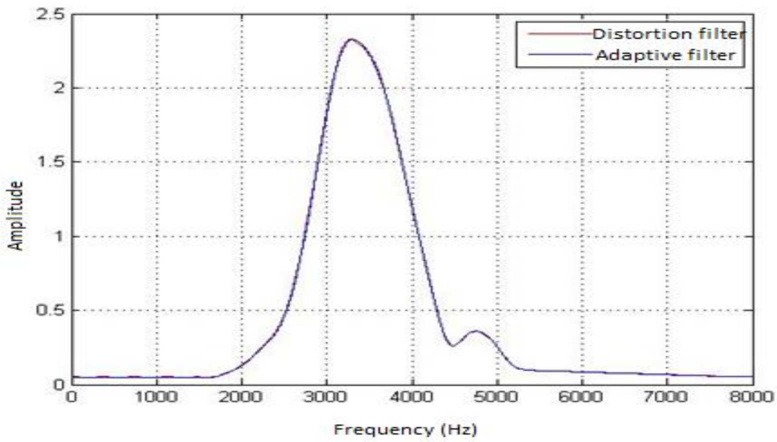

Figure 10. The frequency comparison between the dirtortion filter and the adaptive filter

Frequency response for the IIR filter compared to the frequency response of the distortion filter.

We observed in Figure 10 after analyzing the signals both in time and in frequency and in power that the signal is recovered from the noise by both methods.[9]

5. Conclusions

The experiment simulates the adaptive equalization of some communication channels. In order to illustrate the principle of adaptive equalization, a system of data transmission in the basic band will be considered. In such a system, the data sequence is applied, at emission, to a formation filter, with the purpose of limiting the occupied bandwidth (Low Pass Filterer).

A relatively wide band noise is also introduced on the communication channel. The transmission channel can also be modeled as a linear filter (this is represented by the distortion filter).

To avoid the distortion introduced by the channel, another filter, with the role of

equalizer. This should compensate for the distortion introduced by the channel. The channel transfer function is not generally known, and in addition, it may vary slowly over time. Therefore, the need for this filter appears to be adaptive. To initially perform the adaptation, a training sequence, known at the reception, can be transmitted first and the output of the filter is compared with a locally generated reference signal, but identical to the one emitted. After this initial adaptation phase, the transmitter transmits the information symbols (the useful signal).[8]

Another method of filtering noise using an adaptive filter is to identify the distortions produced and eliminate them with a new filter designed by us. The last method is more précised and although it needs more time, the difference in time is very small so it can be used in telecommunications.

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