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Adaptive Channel Estimation Algorithm For Multi Input Multi Output System – A Brief Review

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Abstract:
Channel estimation plays affective role on the performance of wireless and wire communication systems, since its knowledge is utilized to track and analysis the signal data symbols. Channel estimation is very important technique especially in wireless network systems where the wireless channel change over time, usually caused by transmitter and/or receiver being in motion and/or stable. In order to provide stability and high data receiving rates at the receiver, the system needs an accurate and minimum error estimate of the time-varying channel. Channel estimation is based on the training sequence of bits and which is unique for a certain transmitter and which is repeated in every transmitted burst. The modulated corrupted signal from the channel has to be undergoing the channel estimation using algorithms like Least Mean Squares, Normalized LMS, Variable Step Size LMS, Recursive Least Squares, Least Mean-Squares Newton Algorithm etc are used buffer and estimate the receiver end signal. In this paper a brief reviewing the different channel estimation algorithms for demodulation and comparing all of them with its output performances at various inputs signal and also proposed new algorithm for multi input multi output wireless communication system in noisy and motion environment.

Index Terms— Channel estimation, Multiple Input Multiple Output, Least Mean Squares, Normalized LMS, Variable Step Size LMS, Recursive Least Squares, Least Mean-Squares Newton Algorithm, Mean square error.

Introduction

Channel estimation (CE) is a very important term in communication which plays a vital role in the performance of wireless communication systems, since its knowledge is utilized to detect the data symbols.

Channel estimation is an important technique especially in wireless network systems where the wireless channel changes over time, usually caused by transmitter and/or receiver being in motion. Wireless communication is adversely affected by the multipath interference resulting from reflections from surroundings, such as hills, buildings and other obstacles. In order to provide reliability and high data rates at the receiver, the system needs an accurate estimate of the time-varying channel. Furthermore, wireless systems are one of the main technologies which used to provide services such as data communication, voice, and video with quality of service for both mobile users and nomadic [17]. The knowledge of the impulse response of mobile wireless propagation channels in the estimator is an aid in acquiring important information for testing, designing or planning wireless communication systems [3].

Channel estimation is based on the training sequence of bits and which is unique for a certain transmitter and which is repeated in every transmitted burst. The modulated corrupted signal from the channel has to be undergoing the channel estimation using Least Mean Squares (LMS), Normalized LMS (NLMS), Variable Step Size LMS (VSS LMS), Recursive Least Squares (RLS), Least Mean-Squares Newton (LMSN) Algorithm etc before the demodulation takes place at the receiver side. The channel estimator is shown in figure 1.

In communications especially in wireless communications, interference is anything which modifies, or disrupts a signal as it travels along channel between a source and a receiver. The term typically refers to the addition of unwanted signals to a useful signal.

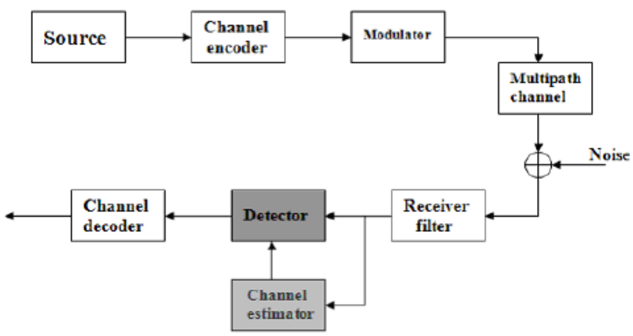


FIG. 1 : THE BLOCK DIAGRAM OF THE CHANNEL ESTIMATOR

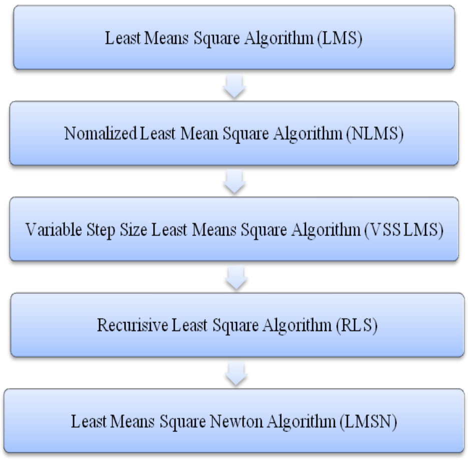
In wireless communication, inter channel interference (ICI) is a form of distortion of a signal in which one symbol interferes with subsequent symbols. This is an unwanted phenomenon as the previous symbols have similar effect as noise, thus making the communication less reliable. ICI is usually caused by multipath propagation or the inherent non-linear frequency response of a channel.

The presence of ICI in the system introduces errors in the decision device at the receiver output. Therefore, in the design of the transmitting and receiving filters, the objective is to minimize the effects of ICI, and thereby deliver the digital data to its destination with the smallest error rate possible.

Multiple input multiple output (MIMO) channels have been introduced to achieve high data speed requisite by the next-generation communication system. The use of the MIMO channels provides higher spectral efficiency versus single input single output (SISO), single input multiple output (SIMO), multiple input single output (MISO) channels, when the available bandwidth is inadequate [8], [13].

I. CURRENT STATUS

Current status in the field of an adaptive algorithm is given below:



II. CHANNEL ESTIMATION ALGORITHMS

Adaptive CE is one the most important current research interests in the wireless communications where the channel is rapidly time-varying. An adaptive algorithm is a process that changes its parameter as it gains more information of its possibly changing environment. This method tries to adjust the filter parameter in such a way that minimizes the Mean square error (MSE) between the output of the filter and the desired signal. Therefore, the adaptive filter parameters are entirely known, replicates the system in question whose parameters are unknown. In other words, the parameters of the adaptive filter give a good approximation of the parameters of the unknown scheme. The performance of this type CE algorithm is dependent on the convergence towards the true channel coefficients, computational complexity as well as minimum MSE performance [14].

A. *Least Mean Squares (LMS) Algorithm*

LMS algorithm has been the most popular as it is simple and effective. However, a limiting factor of LMS is that if the step-size of the algorithm is kept high then the algorithm converges quickly but the resultant error floor is high. Lowering the step-size is results in improvement in the error performance but the speed of the algorithm becomes slow [15]. Least Mean Square (LMS) algorithm used in the area of automatic control, radar, signal processing.

LMS algorithm is given by the following iteration equation:

$$y(n) = WT(n)\,X(n)\,;$$
$$e(n) = d(n) - y(n);$$
$$W(n + 1) = W(n) + 2\mu\, e(n)\, X(n).$$

Where **y (n)** is the output of adaptive filter, **W (n)** is the weight coefficient vector of adaptive filter, **X (n)** is the input vector, **d (n)** is the desired output, **e (n)** is the error signal, and μ is the step-size. LMS algorithm can converge when $0 < \mu < 1/\lambda$, in which $\boldsymbol{\lambda}$ is the maximum Eigen value of input signals' autocorrelation matrix.

In all practical applications, the signals involved might be corrupted by noise. When the noise is present in the received sequence, interference will also in the coefficients adaption process through the error. As a result, where the distribution of the noise is highly impulsive, the LMS scheme might have low convergence and lower steady state MSE performance. The step size parameter, μ determines the convergence rate of the algorithm and higher value provides faster convergence. However, if μ exceeds certain bound then the algorithm will diverge. As the bound on μ is not known a priori and is dependent on the various statistics. In practice, a somewhat conservative scalar value of μ is used. Also a higher value of μ results in higher variations in the tap weight vector estimate after the initial convergence phase. Such variations result in increased distortion in the combined output which in turn results in an increased MSE [7], [15].

B. *Normalized LMS (NLMS) Algorithm*

The main problem of the LMS CE algorithm is that it is sensitive to the scaling of its input signals. This makes it very hard to choose μ that guarantees stability of the algorithm. The NLMS is a variant of the LMS algorithm that solves this problem by normalizing with the power of the input signal. The NLMS algorithm can be summarized as [11] [15].

$$y(n) = WT(n)\,X(n)\,;$$
$$e(n) = d(n) - y(n);$$
$$W(n + 1) = W(n) + 2\mu\, e(n)\, X(n).$$

When a constant scalar step size μ is employed in the LMS/NLMS algorithm, there is a trade off among

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the steady state error-convergence towards the true channel coefficients W(n), which avoids a fast convergence when the step size μ is preferred to be small for small output estimation error e(n). In order to guarantee the algorithm to be convergent, the range of step size μ is specified but the choice of optimal learning step size has not been appropriately addressed. In order to deal with these troubles, one key idea is to exploit varying step size during adaptation.

C. Variable Step Size LMS (VSS LMS) Algorithm

The VSS-LMS algorithm involves one additional step size update equation compared with the standard LMS algorithm. The VSS algorithm is [4], [5], [6]

$$y(n) = WT(n)\, X(n)\, ;$$
$$e(n) = d(n) - y(n);$$
$$\mu(n) = \beta\, (1 - \exp(-\alpha|e(n)|2));$$
$$W(n + 1) = W(n) + 2\mu\, e(n)\, X(n).$$

When the channel is fast time-varying then algorithm cannot accurately measure the autocorrelation between estimation error e(n) to control step size μ update. So, this CE algorithm cannot provide the minimum MSE in the tracking problem, since it cannot acquire and track the optimum step size μ. It may even cause worse steady state results, when the algorithm parameters are not appropriately adjusted. In addition, control parameters α and β need to be adjusted for a better performance. As can be seen here, a general characteristic of these VSS CE methods is that predetermined control parameters are necessary to improve the performance. Though, in most of them, rules to choose control parameters are not specified. Those parameters are always selected from extensive simulations, or from experience. It is clear that the choice of parameters would significantly influence the performance of these schemes [12].

Various Variable step-sizes (VSS) based LMS have a high step-size initially for fast convergence but then reduce the step-size with time in order to achieve a low error performance. All VSS algorithms aim to improve performance at the cost of computational complexity. This trade-off is generally acceptable due to the improvement in performance. [10], [16], [20]

Several variable step-size algorithms designed to enhance the performance of the LMS algorithm have been given. However, the algorithms in are very sensitive to interference noise, while the method in needs the noise signal to be uncorrelated, and the method proposed in is only suitable for stationary and low-level noise conditions; thus, they are limited in many applications. To the best of our knowledge, no variable step-size LMS algorithm has been proposed for a wide range of applications where the noise signal is correlated, potentially high variance, such as speech signals. [1], [2]

D. Recursive Least Squares (RLS) Algorithm

To combat the channel dynamics, the RLS based CE algorithm is frequently used for rapid convergence and improved MSE performance [18]. The standard RLS algorithm is

$$y(n) = WT(n)\, X(n)\, ;$$
$$e(n) = d(n) - y(n);$$
$$W(n + 1) = W(n) + 2\mu\, e(n)\, X(n).$$

Where l is the exponential forgetting factor with 0 < μ < 1. The smaller value of l leads to faster convergence rate as well as larger fluctuations in the weight signal after the initial convergence. On the other hand, too small l value makes this algorithm unstable. Subsequently, it requires best possible forgetting factor such that the estimator error is decreased. Although a lot of modified CE algorithm has been studied on employing adaptive forgetting factor and parallel forgetting factor, the CE performance is severely degraded in highly dynamic fading channel even when the forgetting factor is well optimized [15]. However, this scheme also has computational complexity-performance trade off problem that is the major obstacle for practical mobile terminal as well as base station (BS) implementation [9], [18]. Consequently, an efficient CE algorithm better than existing algorithms is required which gives both fast convergence and minimum steady state MSE.

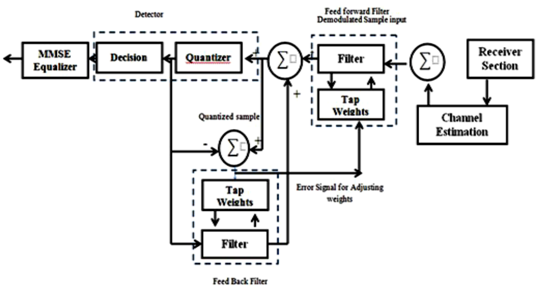

FIGURE 2: BLOCK DIAGRAM OF RECURSIVE SECTION

E. Least Mean-Squares Newton (LMSN) Algorithm:

LMSN is known to outperform the standard LMS algorithm when the data use as a large Eigen values spread or when the regressor matrix involved are in conditions such as is the case with the memory polynomial model. This algorithm exhibits many desirable characteristics such as stability, robustness and accuracy. LMSN algorithm used less parameter than the RLS algorithm, which utilized forgetting factors. This algorithm depended on a user selected constant is removed by dropping the parameter. [19]

For n = 1 : N

$$R\text{-}1(0) = \delta I$$
$$W(0) = [\, 0 \,....\, 0\,]T$$

for n = 1 : N

e(n) = d(n) – y(n)

$R^{-1}(n) = R^{-1}(n-1) - \frac{R^{-1}(n-1)\, xH(n)\, x(n)\, R^{-1}(n-1)}{1+ x(n)\, R^{-1}(n-1)\, xH(n)}$

W(n) = W(n-1) + μ R-1(n) xH(n) e(n)

Where

δ is a constant usually having large values (≥ 103 in this study).

μ is the step size, where $0 < \mu < 1$

R-1(n) is the modified input variable which is affect by x(n)

III. PROPOSED INPUT PARAMETERS

A. WCDMA (Wideband Code Division Multiple Access)

Wideband CDMA is a third-generation (3G) wireless standard which utilizes one 5 MHz channel for both voice and data, initially offering data speeds up to 384 Kbps. WCDMA was the 3G technology used in the US by AT&T and T-Mobile. W-CDMA can support mobile/portable voice, images, data, and video communications at up to 2 Mbps (local area access) or 384 Kbps (wide area access). The input signals are digitized and transmitted in coded, spread-spectrum mode over a broad range of frequencies. A 5 MHz-wide carrier is used, compared with 200 KHz-wide carrier for narrowband CDMA.

B. Wireless Internet

Wi-Fi is a wireless networking technology that allows computers and other devices to communicate over a wireless signal. It describes network components that are based on one of the 802.11 standards developed by the IEEE and adopted by the Wi-Fi Alliance. Examples of Wi-Fi standards, in chronological order, include:

802.11a, 802.11b, 802.11g, 802.11n, 802.11ac, 802.1x

Wi-Fi is a local area wireless technology that allows an electronic device to participate in computer networking using 2.4 GHz UHF and 5 GHz SHF ISM radio bands.

C. DTS Signal

DTS is a series of multichannel audio technologies owned by DTS, Inc. (formerly known as Digital Theater Systems, Inc.), an American company specializing in digital surround sound formats used for both commercial/theatrical and consumer grade applications. It was known as The Digital Experience until 1995.

IV. ANALYSIS OUTPUT PARAMETERS

A. Noise

In signal processing, noise is a general term for unwanted (and, in general, unknown modifications that a signal may suffer during capture, storage, transmission, processing, or conversion.

Sometimes the word is also used to mean signals that are random (unpredictable) and carry no useful information; even if they are not interfering with other signals or may have been introduced intentionally, as in comfort noise.

Noise reduction, the recovery of the original signal from the noise-corrupted one, is a very common goal in the design of signal processing systems, especially filters.

Noise is unwanted electrical or electromagnetic energy that degrades the quality of signals and data. Noise occurs in digital and analog systems, and can affect files and communications of all types, including text, programs, images, audio, and telemetry.

B. Mean Squared Error (MSE)

In statistics, the mean squared error (MSE) of an estimator measures the average of the squares of the "errors", that is the difference between the estimator and what is estimated. MSE is a risk function, corresponding to the expected value of the squared error loss or quadratic loss. The difference occurs because of randomness or because the estimator doesn't account for information that could produce a more accurate estimate.

C. Response Time

The convergence time is one of the key factors which determines performance of the dynamic routing protocol.

COMPARISON BETWEEN CE ALGORITHMS

TABLE 1: COMPARISON BETWEEN CE ALGORITHMS

S. No.	Algorithm	Output Parameters			
		Computational complexity	MSE & Noise	Range	Response Time
1	LMS	Low	High	Low	High
2	NLMS	High	High	Low	Low
3	VSS LMS	Medium	Medium	Medium	Medium
4	RLS	High	Low	High	Medium
5	LMSN	Low	Medium	High	High

V. PROPOSED ALGORITHM

The LMSN and LMS algorithm is used a slightly simplified version of the Recursive least squares (RLS) algorithm that requires a lower number of computations in addition to being more numerically robust. An additional advantage of the LMSN algorithm is that it uses less parameter than the RLS algorithm, which utilizes a forgetting factor.

The LMSN algorithm was found to outperform the RLS algorithm in terms of both estimation accuracy and speed. The LMSN adaptive filtering algorithm is shown to consistently outperform the RLS algorithm by 1 dB in terms of error performance while requiring on average 10% less computational time.

Expected outcome to proposed algorithm:

- ✓ It is required less input parameter to send the signal.
- ✓ It is combine noise and input signal in different step.
- ✓ It is less computational complexity.
- ✓ It is give low mean signal error.

VI. COMPARISON

Comparing all of them with its input and output performances. Also proposed a suitable algorithm for multi input multi output (MIMO) system in noisy environment shown in table 1.

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