LOW COMPLEXITY BLIND ADAPTIVE CHANNEL SHORTENING FOR MULTICARRIER COMMUNICATION SYSTEM

by

Muhammad Yamin

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Muhammad Yamin (205-FET/MSEE/F08)

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International Islamic University, (IIU), Islamabad.

Supervisor: Dr. Rab Nawaz

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In the name of Allah (SWT) the most benificient and the most merciful.

CERTIFICATE

Title of Thesis: **<u>Low Complexity Blind Adaptive Channel Shortening For Multicarrier Communication System</u>**

Name of Student: <u>Muhammad Yamin</u>

Registration No: <u>205-FET/MSEE/F08</u>

Accepted by the Faculty of Engineering and Technology, International Islamic University Islamabad, in partial fulfilment of the requirements for the Master of Philosophy Degree in Electronic Engineering.

Viva voice committee

<u>External Examiner</u>

<u>Internal Examiner</u>

<u>Supervisor</u>

Dr. Rab Nawaz
Adjunct Faculty,
FET, IIU, Islamabad

10 August 2011

Declaration

I certify that except where due acknowledgments has been made, the work has not been submitted previously, in whole, to qualify for any other academic award, the content of the thesis is the result of work which has been carried out since the official commencement date of the approved research program, and any editorial work paid or unpaid, carried out by a third party is acknowledged.

Muhammad Yamin

205–FET/MSEE/F08

DEDICATED TO

THE HOLY PROPHET (P.B.U.H)

THE GREATEST SOCIAL REFORMER

AND AFTERWARDS

TO MY WORTHY PARENTS WHO HAD WISHED ME WITH THIS DISSERTATION

ABSTRACT

The purpose of channel shortening is to condense the channel in a shorter span to make the Multi Carrier Communication Systems bandwidth and power efficient. The autocorrelation minimization based channel shortening algorithms are investigated in this thesis. It is well known in stochastic literature that if a white signal is input to a filter having a short span, the autocorrelation introduced in the output signal is small. The SAM algorithm expects that the reverse might also be true. Therefore, it performs channel shortening by minimizing the sum-squared autocorrelation of the output of the channel for a range of lags. We demonstrate that minimizing the autocorrelation of the channel output as in SAM is not equivalent to condensing the channel in a contagious window. We show that as long as the ADSL channels are concerned, identical channel shortening can be achieved by using a single autocorrelation in the cost function. We also elaborate that this single autocorrelation is not specific to any particular value of lag. Using a range of autocorrelations is unintelligent and overkill. This finding reduces the computational complexity of channel shortening. We name our algorithm Any Lag Autocorrelation Minimization (ALAM). The simulations support the ideas presented in the thesis. We explain reasons behind the diverging behaviour of autocorrelation based algorithms to shorten the ADSL channels.

The thesis serves to identify lags which can be minimized in the true sense of channel shortening. It is found that these lag values are not driven by the nature of the impulse response of the underlying channel. They always try to shorten a channel around its

mid point. Therefore, it is conjectured that if the channel has its mass of energy around its mid point, ALAM is flexible and will successfully shorten it to different window lengths without any diverging behaviour. On the other hand, SAM, which minimizes all of the channel taps in its cost function, is expected to fail in such situations. The failure will come in the form of no shortening at all OR divergence from the optimal point.

PUBLICATIONS

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May Allah Bless you all with endless happiness!

ACRONYMS

ALAM	Any Lag Auto Correlation Minimization
AWGN	Additive White Gaussian Noise
AR	Auto Regressive Model
ADSL	Asymmetric Digital Subscriber Line
BEP	Bite Error Probability
BER	Bit Error rate
CDMA	Code Division Multiple Access
CFO	Carrier Frequency Offset
CMA	Constant Modulus Algorithm
CNA	Carrier Nulling Algorithm
CSA Loop	Carrier Serving Area Test Loop
CP	Cyclic Prefix
CNR	Carrier to Noise ratio
DAB	Digital Audio Broadcast
DVB	Digital Video Broad Cost

DD	Decision Directed
DFT	Discrete Fourier Transform
DMT	Discrete Multitone Transceiver
EC	Echo Canceller
FFT	Fast Fourier Transform
FET	Frequency Domain Equalizer
FIR	Finite Impulse Response
GA	Genetic Algorithm
HOS	Higher Order Statistics
IID	Independent Identical Distribution
IFFT	Inverse Fourier Transform
ISI	Intersymbol Interference
ICI	Inter Channel Interference
IBI	Inter Block Interference
LS	Least Square Algorithm
LMS	Least Square Algorithm
MA	Moving Average Model
MSE	Mean Square Error
MMSE	Minimum Mean Square Error

MERRY	Multicarrier Equalization by Restoration of Redundancy
MIMO	Multiple Input Multiple Output
MLSE	Maximum Likely hood Sequence Estimation
MBR	Maximum Bite Rate
NLMS	Normalized Least Mean Square Error
OFDM	Orthogonal frequency Division Multiplexing
PTEQ	per Tone Equalization
SIMO	Single Input Single Output
SISO	Single Input Single Output
SNR	Signal to Noise Ratio
SAM	Sum Square Auto Correlation Minimization
SAAM	Sum Square Absolute Auto Correlation Minimization
SLAM	Single Lag Autocorrelation Minimization
SSNR	Shortening Signal to Noise Ratio
TEQ	Time Domain Equalizer
VDSL	Very High Speed DSL
WSS	Wide Sense Stat

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Chapter 1.

PRINCIPLE OF CHANNEL SHORTENING

1.1. Introduction

Equalizer is a compensator that counters the effects of Inter Symbol Interference (ISI), which source to higher error rates for a single carrier transmission system. Inter Symbol Interference (ISI) is due to frequency selective characteristics of the communication channel. The equalization process is employed at receiver for interference rectification in response to delay spread of the channel. Likely, in some cases, various intelligent techniques are being deployed at the transmitter side to make a signal less prone to delay spread.

In Multi Carrier (MC) communication systems, a "cyclic prefix" is inserted to the start of each symbol to convert linear convolution (between transmitted symbols and the channel impulse response) into circular convolution. The Cyclic prefix (CP) should be equal in length to the expected delay spread of the channel. The demodulation is performed at the receiver by discarding the starting part of the received symbol equal to the cyclic prefix length. The remaining part is fed to an FFT block. This whole arrangement makes the MC system ISI free and simplifies its equalization to a simple point wise division of the FFT output. The transmission of the CP reduces bandwidth and power efficiency. If the expected delay spread of the channel is large, CP would be incremented to a considerable length. What amendments should be done in receiver to kill this incremental in CP length and bounds it to a predefined length.Therefore,an equalizer is designed in such a way that the convolution of channel and equalizer generate an effective channel , shortened to this length. This method/approach is known as channel shortening and the equalizer is referred to as the Time Domain Equalizer (TEQ).

Channel shortening works on the same principles like equalization. Blind equalizers require algorithms to update tape coefficient for untrained sequence. Adaptive equalizers on the other hand are good for time varying channels. Presently, majority of the channel shortening literature is related to the ADSL systems because the expected channel delay spreads are large. Recently this communications system has received much attention on account of its low complexity and excellent performance compared to traditional modem systems. Asynchronous Digital Subscriber Line (ADSL), have become very popular and bring a great revolution telephone in wire line communications. This report highlights the need for channel shortening and then proposes an efficient technique to reduce complexity in existing blind, adaptive TEQ design algorithms. Moreover, the notations used through out the thesis would be established in the continuation of this chapter.

1.2. Multi Carrier Modulation

Multicarrier is actually be the modulation technique deployed to any type of communication access technique like multiuser, multichannel, multiple input multiple Output. For all, MC has to combat channel dispersion and to counteract with delay spread of the channel. In many cases receiver has to jointly shorten multiple channel (MC-CDMA) using a single TEQ. For MC-CDMA system, multiple users each spread their signal by means of a spreading code prior to acquire multicarrier modulation [46].To boost up performance, in receiver, all the user's channel can be jointly shortened, to mitigate ISI prior to dispreading take place. Block diagram is only for the system which has only a single input and single output but can also be designed for MIMO streams henceforth.

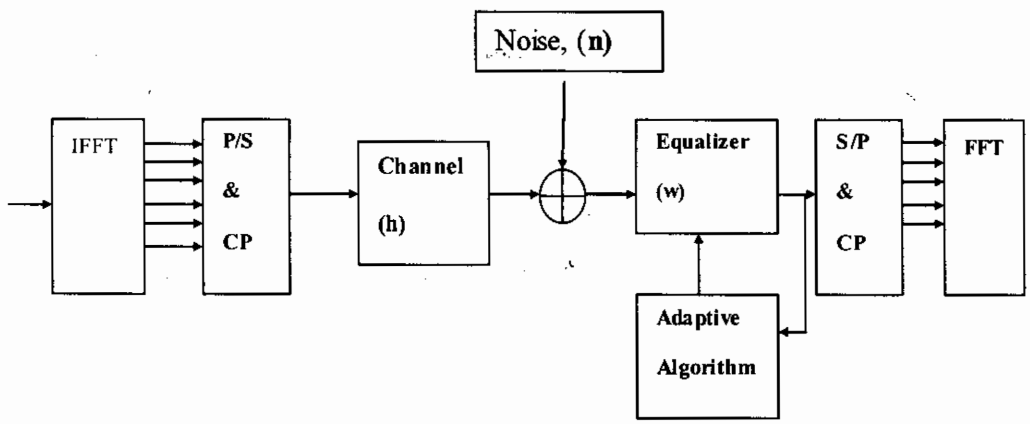

Figure 1.A Multi Carrier Communication System.

Multicarrier based system like OFDM and ADSL has been gaining popularity over recent year and extensively used in different transmission system like Digital Video Broadcast (DVB), Digital Audio Broadcast (DAB), Wireless Local Area Network (WLAN), and Digital Subscriber Line (DSL). The multi carrier modulation technique divides the available bandwidth among independent, orthogonal and parallel sub channels/subcarriers. Because of orthogonality between subcarriers the data can be sent in parallel in an addition to the cyclic prefix to avoid any ISI and inter carrier interference (ICI). If CP length is not long enough to be equal to the delay spread of the channel, orthogonality of subcarriers is lost, which causes ISI/ICI [1].

To mitigate the effect of ISI/ICI between subchannels, a short time domain filter (TEQ) is typically placed at the receiver in an advance to FFT Operation. The TEQ is usually a finite impulse response (FIR) filter whose purpose is to shorten the impulse response of the channel as normally being effected in response to the delay spread of the channel. Single input and single output system model is shown in the above figure. To illustrate SISO based multi carrier system, the bite stream is first divided into N sub streams by means of Serial to parallel converter. Each sub stream is linearly modulated (QAM/PSK) comparative to its subcarrier frequency. IFFT is an efficient technique for smooth implementation of discrete time modulation. A CP is appended to each symbol to ensure the orthogonality between consecutive sub channels [1].

The receiver discards CP in the received signal and remaining bins are then processed by the receiver. In order for the subcarrier to be independent, the convolution of the signal and channel would be circular. This cyclic prefix essentially transform linear convolution into cyclic convolution. If CP length is equal to channel than the output of each sub channel at the FFT output would be its input multiplied by a scalar gain factor. The signal in the sub channels can than be equalized by pool of scalars referred to as Frequency Domain Equalizer (FEQ). It should be highlighted that even after designing TEQ, the design of FEQ remains efficient.

1.3. Organization of Thesis

The remaining potion of this dissertation is organized as follow.

Chapter2 organized and explore the previous "TEQ" designed method for multicarrier system, employed in both the fixed (DMT) as well as wireless (OFDM) network. In particular, this chapter describes channel shortening techniques like MSSNR [2], SAM [3], SLAM [4], LHSAM [5], and GLHSAM [6].It covers the entire essential mathematical details and principle.

Chapter.3 detailed the motivation and principle structure of proposed (ALAM) algorithm.ALAM works on the principle of minimizing the square value of auto correlation of out put signal by selecting an appropriate lag. It gives an insight into the channel and incorporates a universal technique for lag selection. The proposed algorithm achieved channel shortening in true spirit and more suitable for the case when channel varying fast.ALAM is more stringent to provide flexibility in lag selection according to the variation in channel impulse response and hence to improve drop down in output bite rate as being a crucial problem in all viewed technique.

Chapter.4 The performance assessment is evaluated through simulation in Matlab and results are presented for CSA Loop ADSL channels. Enough simulation has been given to verify the validity as proposed in chapter 3 for ALAM Algorithm. Comments about the simulation results are made.

Chapter .5 concludes the dissertation. It gives the summery of the results as obtained through "ALAM".Conclusions are drawn and future directions are suggested.

Chapter 2.

LITERATURE SURVEY

This chapter will discuss the multitude of multicarrier equalizer design in the literature. But the focus will be mainly to those directed towards the designed technique as presented in this dissertation. Section 2.1 set up a common trained method to designed formulation on the basis of maximizing the shortening SNR. MSSNR is the most versatile and optimum techniques that have a window placement control in trained method. Section 2.2 formulate a vide concept over Blind designed technique. Section 2.3, 2.4 and 2.5 detailed further variance of Blind Channel shortening Technique.Therefore,a more directed channel shortening algorithms relevant to this dissertation are briefly reviewed in this chapter.

2.1. Maximum Shortening SNR (MSSNR) Method

Maximum Shortening SNR (MSSNR) method was proposed by Melsa et al, in [2]. MSSNR is based upon minimizing the energy of the channel outside the target window while keeping the energy inside constant. Usually, it is not possible to shorten the impulse response perfectly. A few energy taps will lie on the outer side of the window containing the largest (v+1) consecutive samples of the effective channel. What can be done is to force as much of the effective channel's impulse response as possible to lie in v+1 consecutive samples. The MSSNR TEQ design method exploits the ratio of the effective channel impulse response energy inside a target window of length v+1 samples to the energy outside window of the channel. With reference to Figure (1.2), the effective channel impulse response is given by

$$c = h * w \tag{2.1.1}$$

Where, * denotes linear convolution. The shape of impulse response of the effective channel [c] inside the (v+1) length window is usually unimportant; what is important is that the SSNR be maximized. The MSSNR channel shortening is shown in Figure 2

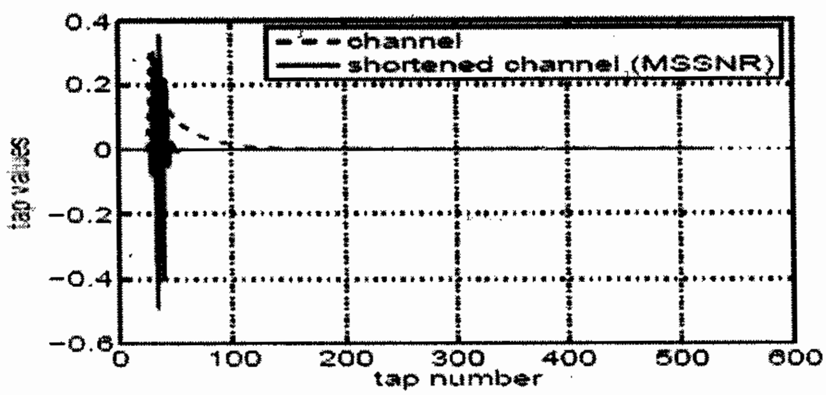

Figure 2. MSSNR Fundamental Channel Impulse Response

Figure.'2' shows the impulse response of "Original CSA Loop1" Channel and MSSNR based shortened channel.Unshortened channel carries energy in 200 consecutive samples where as the shortened channel have the same energy in 33 consecutive taps.MSSNR design directly deals with the concept of channel shortening. It is because; it operates on the channel with the gal of shortening it without regard to any other performance matrices. Since MSSNR method employed get triggered on the basis of ratio of energy between shortened and unshortened channel.Therefor next section describe its structure according their relative energy ratio by limiting the boundary.

Energy Ratio (Inside and outside)

MSSNR "TEQ" attempts to maximize the ratio of energy in a window of effective channel and energy out side the window of effective channel.Figure."2" indicates the position of inside (Win) and outside (Wall) and their corresponding energies lies to their corresponding position.

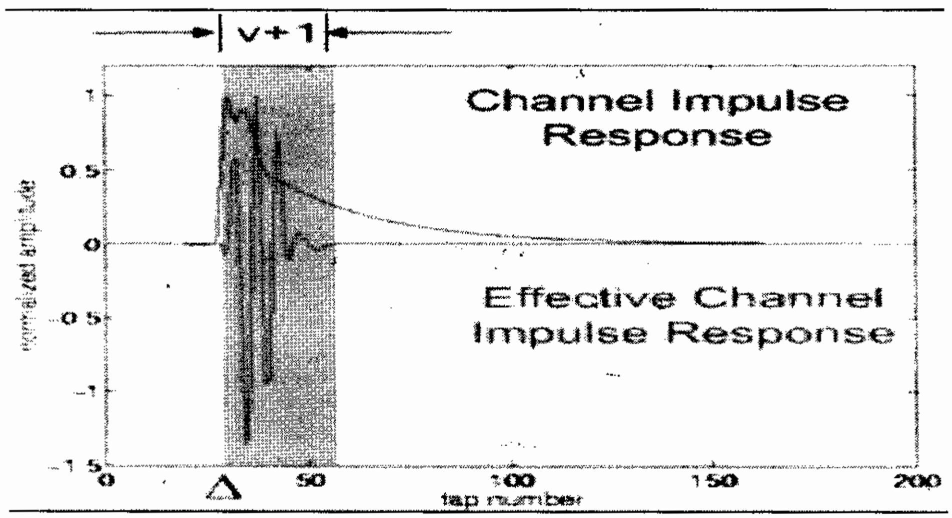

Figure 3. MSSNR based channel shortening [2].

MSSNR TEQ maximizes energy inside the target window and minimizes out side the target window. This relation is best described by using the following equation.

$$\max_{w}(SSNR) = \max_{w} 10\log_{10}\left(\frac{\text{energy inside window after TEQ}}{\text{energy outside window after TEQ}}\right)$$

If $\mathbf{H}$ denotes the convolution matrix of the original channel $\mathbf{h}$, then the effective channel will be:

$$c = H w \qquad (2.1.2)$$

Since the effective channel has been portioned into two parts. So the part of channel samples lying within the desired (v +1) window are denoted by

$$c_{win} = H_{win}\, w \qquad (2.1.3)$$

And the part of channel samples lying outside the desired $(v+1)$ window are denoted

by $$c_{wall} = H_{wall}\, w \qquad (2.1.4)$$

And corresponding Hwin and H_{wal}l matrixes are given as:

$$\mathbf{H}_{win} = \begin{pmatrix} h(\Delta) & \cdots & h(\Delta - Lw + 1) \\ \vdots & \ddots & \vdots \\ h(\Delta + v) & \cdots & h(\Delta + v - Lw + 1) \end{pmatrix}$$

$$\mathbf{H}_{wall} = \begin{pmatrix} h(0) & 0 \ldots\ldots\ldots\ldots & 0 \\ h(\Delta + v + 1) & h(\Delta + v) & h(\Delta + v + 1) \\ h(0) \ldots\ldots\ldots\ldots & 0 & 0 \end{pmatrix}$$

Here, $\mathbf{H}_{win}$ consists of v+1 rows of **H** starting from some Δth row, and H_{wall} consists of the remaining rows of H. The MSSNR design is based on the following equation:

$$w^{opt} = \arg\max_{w} \prod_{j=1}^{M} \frac{w^T B_j w}{w^T A_j w} \qquad (2.1.5)$$

Where **A** and **B** are given by

$$A = H_{wall}^{H} H_{wall} \qquad (2.1.6)$$

$$B = H_{win}^{H} H_{win}$$

Therefore the objective cost function will be

$$\max_{w}(SSNR) = \max_{w} 10 \log_{10}\left(\frac{w^T B w}{w^T A w}\right) \text{ subject to } w^T B w = 1$$

Effectively, shortening is carried out by minimizing the wall energy (the denominator) by keeping the window energy (the numerator) equal to unity. The performance metric for channel shortening is ratio of the energy of the shortened channel inside a window of length (v+1) to its energy outside that window. It is called Shortening Signal to Noise Ratio (SSNR). The algorithm makes matrices A and B at different possible Δ, and designs the filter at the Δ where the SSNR metric is maximized.

When the lengths of w goes above the length of the cyclic prefix, the matrix B turn into singular and its inverse does not exist. In [7] it was suggested to maximize the energy inside the window i.e., $w^T Bw$ while keeping the energy outside the window i.e. $w^T Aw$ equivalent to unity. The matrix A is always positive definite and a longer length TEQ may be considered to obtain performance gains. The MSSNR technique requires the knowledge of the channel and it does not take into account the noise involvement in the channel. Its computational complexity is also high as it has to find the best Δ for the channel, to place the v+1 length window.

2.2. Sum Squared Autocorrelation Minimization (SAM) Method

SAM is a blind adaptive channel shortening algorithm to be based upon the correlation estimates of the output signal. It attempts to minimize the auto correlation of output data beyond a window of desired length **v**. The proposed algorithm, to be known as "sum-squared auto-correlation minimization" (SAM), employs the source sequence to be zero-mean, white, and wide-sense stationary, and it is formulated as a stochastic gradient descent algorithm.

The system model is shown in Figure 4 Here x(n) is the source sequence to be transmitted through a linear Finite-Impulse-Response (FIR) channel of length L_h+1 taps. Let r(n) be the received signal, which will be filtered through an L_w+1 tap TEQ, **w**, with an impulse response vector to acquire the output sequence. Here **c** with length L_c+1 is used to denote the effective channel-equalizer impulse response vector

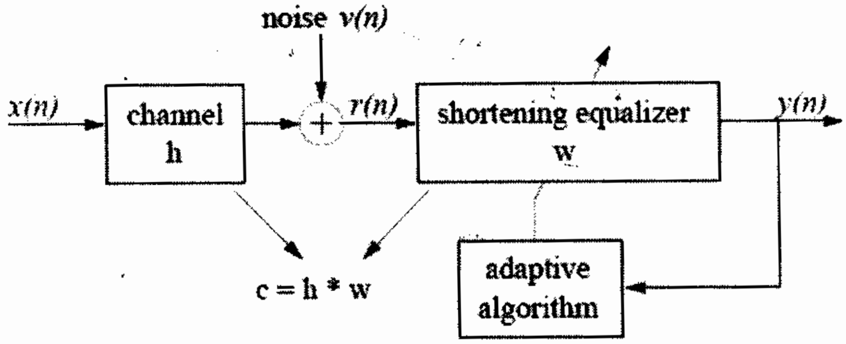

Figure 4. SAM Channel Shortening Model

.Input Received Signal:

$$r(n)=\sum_{k=0}^{Lh}h(k)x(n-k)+v(n) \tag{2.2.1}$$

Equalizer Output Signal:

$$y(k)=\sum_{k=0}^{Lw}w(k)r(n-k) \tag{2.2.2}$$

$$= \mathbf{w}^T\mathbf{r}(n)$$

Where r (n) =[r (n-1), r (n-2), r (n-3)...r (n-Lw)]^T. The following assumptions are made [3].

- The source sequence x(n) is white, zero-mean and wide-sense stationary (W.S.S).
- The relation 2Lc < N holds for multicarrier systems, i.e. the effective channel has length less than half the FFT size.
- The source sequence x(k) has unit variance: $\sigma^2_x=1$.
- The noise sequence n(n) is zero-mean, i.i.d., uncorrelated to the source sequence, and has variance σ^2_n.

Consider the autocorrelation of the effective channel, **c**

$$Rcc(l) = \sum_{k=0}^{Lc} c(k)c(k-l) \qquad (2.2.3)$$

If the effective channel **c** had v+1 non-zero taps, the autocorrelation values $R_{cc}(l)$ would be zeros outside a window of length (2v+1),

$$Rcc(l) = \sum_{k=0}^{Lc} c(k)c(k-l) = 0, \forall\, |l| > \nu$$

Hence, Balakrishnan et al., [3] proposed to shorten the channel by minimizing its autocorrelation outside a window of length (2v+1). Their cost function is defined

$$J_{SAM} = \sum_{l=\nu+1}^{Lc} |Rcc(l)|^2 \approx \sum_{l=\nu+1}^{Lc} |Ryy(l)|^2 \qquad 2.2.4$$

The authors in [3] carry on proving that given the above stated 4 conditions are true,

$$\sum_{k=\nu+1}^{Lc} |Ryy(l)|^2 = \sum_{k=\nu+1}^{Lc} |Rcc(l)|^2 + 2\sigma_v^2 \sum_{k=\nu+1}^{Lw} Rcc(l)Rww(l) + \sigma_v^4 \sum_{k=\nu+1}^{Lc} |Rww(l)|^2$$

Therefore, Output auto correlation will be

$$Ryy(l) = Rcc(l) + \sigma_v^2 Rww(l) \qquad (2.2.5)$$

This equation is also described in [2] and is mentioned in stochastic processes literature where filtering of random signals are governed by the system model shown in Figure 2.2 [8]. Therefore, we can assume that for noiseless scenarios,

$$R_{yy}(l) \cong R_{cc}(l)$$

The cost function is modified as

$$J_{SAM} = \sum_{l=\nu+1}^{L_c} |R_{yy}(l)|^2 \qquad (2.2.6)$$

A stochastic gradient blind adaptive channel shortening algorithm is, therefore, proposed in [SAM] which minimizes the above cost function. It is blind because it does not require any training sequences to be transmitted. Though to calculate L_c, it requires an estimate of the length of the channel **h**. In the case of ADSL, L_c is normally modelled as 512 [1].

SAM algorithm can be better phrased as below; The autocorrelation of the output $R_{yy}(l)$ signal is governed by the span of the channel **c**, provided that the input to the channel is a white signal. Therefore, a channel (filter) **c** having span of v+1 won't introduce autocorrelation in the output for lags greater than v. Hence their cost function is given by the above Equation.

2.3. Single Lag Auto Correlation Minimization (SLAM) Method

Nawaz et al. [4] remarked on the same analogy that if the channel **c** has span greater than or equal to v+1, the output autocorrelation at lag, l=v+1 will be non-zero. It will only be zero if the channel has a span less than v+1. Therefore, a computationally less expensive cost function can be

$$J_{SLAM} = Ryy(l) \quad ,l = v + 1 \tag{2.3.1}$$

Hence on the basis of this optimization of Auto correlation at lag l, it is known Single Lag Autocorrelation Minimization (SLAM). It achieves identical channel shortening like SAM, as will be shown in chapter 4.The SLAM algorithm also introduced a stopping criterion to freeze the adaptation. To keep the present discussion on track, Further discussion concluding these feature would be discussed in the next chapter. The complexity of SLAM is approximately 1/500 times that of SAM for ADSL downstream transmission environment parameters [4]. All the algorithms discussed next use a single lag at each iteration of the algorithm and hence have similar complexity to that of SLAM.

2.4. Lag Hopping Sum Squared Auto Correlation Minimization Method (LHSAM)

Grira et al [5] proposed this view to use, a single lag at each iteration, but hopped randomly between all lags of SAM algorithm. In this way, they suggest to visit all of the lags of SAM algorithm. LHSAM also achieves same channel shortening as SAM and SLAM. LHSAM is an extension as being taken place between SAM and SLAM. It basically used the SLAM cost and reformulate it in such a way that it can provide maximum SIR. For this it adopted its auto correlation lag in range associated with their occurrence probability. LHSAM was the reaction in response of recent analytical results by Walsh, Martin and Johnson as concluded that optimizing the single lag autocorrelation minimization (SLAM) cost gave no guarantee to convergence to high signal to interference ratio (SIR), as an important metric in channel shortening applications.

2.5. Generalized Lag Hopping Sum Squared Auto Correlation Minimization (GLHSAM)

Khaled et al [6] proposed to use, at each iteration a single lag, but hopped between all lags of SAM algorithm. The hopping is not random as in [5] but exponential. On the average, the lags close to the lower limit of SAM lags i.e. v+1 are chosen more than those close to the higher limit of SAM lags i.e. L_c. Their algorithm also achieves same channel shortening as SAM, SLAM and LHSAM.

Chapter 3.

ANY LAG AUTOCORRELATION MINIMIZATION (ALAM)

ALAM like SAM attempts to minimize the sum squared value of auto correlation terms of effective channel impulse response outside a window of desired length. This minimal value of auto correlation can be achieved by selecting Any single lag with in channel length (0~Lc)

3.1. Motivation towards ALAM

ALAM is basically extending the work of SLAM in context to improve SIR, not to move it towards SAM as did LHSAM. But in ALAM, the primary perception is that any best fitted lag can be selected to compute auto correlation between samples as in[6] or between averaged estimated like in SLAM instead to fix it to a single pacific lag 33. So By the virtue of ALAM,a best fitted lag can be approached, which can provide high SIR and to get significant improvement in out put bite rate. This chapter detailed the working principle, Designed model, and TEQ algorithm to be based upon ALAM.

3.2. Basic Principle

SAM works on this principle that If the effective channel ($c = h * w$) is short, its auto-correlation should also be short and reverse might be true. Therefore it performs channel shortening by minimizing the sum squared auto correlation of the out put of the channel for a range of lag. But in ALAM, on the basis of stochastic theory we have argued that auto correlation of a output signal does not reflect its span in time and widely discussed in next section. This is important as it gives an insight into the channel shortening based upon the auto correlation minimization. We also use the instantaneous estimate of auto correlation as in [6].

3.3. Structure of ALAM

If the delay spread of the channel is large, the length of CP is bounded to a proficient length and an equalizer is designed in such a way that the convolution of channel and equalizer generate an effective channel that is shortened to that length. This method is well known as **channel shortening** and equalizer is referred to as **Time Doman Equalizer.** Refer to figure "1",for high SNR scenarios

$$Rcc(l) = \sum_{k=0}^{Lc} c(k)c(k-l) \tag{3.3.1}$$

We recall that the purpose of channel shortening as to condense the effective channel to a contagious window of length (v+1).When we minimize $R_{yy}(l)$ for any given lag l, have there any precise control over minimization?.Which part of the channel we have to minized and which one have to saved from getting minimized.Generally speaking,is there any window placement control in blind channel shortening.But on another hand, MSSNR , being a trained method of channel shortening, had such a control as mentioned in chapter 2.

ADSL CSA Loop channels are modelled as length of 512. In the simulations, all algorithms mentioned above use an equalizer of length 16. Therefore, we have an effective channel of length 527.Keeping an eye on Equation 3.1.1 above, we can easily see that selecting a lag equal $l = 280$ to will exclude the channel taps from tap index 247 to 279 in the $R_{yy}(280)$ cost function. Therefore, we could say that an algorithm using a lag of 280 will minimize all the channel taps except between tap 247 and 279. Therefore, this window of channel taps would be equal to 33. We can appreciate that increasing the lag from 280 will increase the window length from 33 and vice versa. The window position will always be on centre position at tap 264, as the mid point between index 0 and index 526. Hence we have achieved a window control over autocorrelation minimization based channel shortening. Note this choice of lag cannot be smaller than 264. For lag $l = 264$, we would have been minimized all of the channel taps. Therefore, one legitimate range of lags for SAM could be from 280 to Lc. This will effectively put more pressure on the starting and ending taps with decreasing pressure towars the centeral taps. The window control is unluckily not driven by the nature of the channel but by the nature of the autocorrelation. The autocorrelation will/can only provide us a window at the midpoint of channel taps indexing from zero to L_c. For MSSNR, the placement of the window is governed by the channel. In our simulations ADSL channels for MSSNR method, the optimal window is always placed where the major portion of the non-zero taps of the CSA loop channel reside. The CSA test loops typically have almost all of their energy in 200 consecutive taps and typically become very small after tap index 230 onward. These taps do not lie in the range 247 to 279. Our simulations show that if we dictate MSSNR method to put its window in the range 247 to 279, it completely fails to shorten the ADSL channels.

Let us now revisit SLAM algorithm which suggests to use lag, $l = 33$ Equation 3.1 suggests that the channel taps which will be minimized are from tap index 0 to 526. But a careful examination tells us that the taps between index 33 and 493 will be included in minimization twice. Does it in any way support channel shortening. All we are doing is that we are minimizing the *whole* channel with putting more emphasis on the middle part.

Let us revisit SAM algorithm which suggest to use all lags in the range $l = 33$ *to* $l = Lc$ As long as the lag is between $l = 33$ *to* $l = 280$,all taps are being included in minimization but we have increasing minimization towards the center of taps between index 33 and 493. For lags greater than 264,as explained before, we have now increasing minimization on the outer taps centering at 264. In a nutshell, we are unable to provide a safety to some range of taps from being minimized to achieve channel shortening. Interstingly, Ishaq et. al [9] propose to use lag $l = 1$Their channel shortening is also indetical to SAM, SLAM, LHSAM, Khaled et al algorithms.

All this discussion leads to the conclusion that the autocorrelation minimization based upon channel shortening should be looked at a new angle. All the algorithms keep on minimizing all the taps of the channel without any regard to keeping some taps away from minimizing. If we look at the shortened channel by one of these algorithms, we notice that the shortened channel has norm even smaller than that of the original channel. The algorithms during updation reach a point where somehow, the energy of the channel in some window of length v+1 is maximized over the energy in the remainder of the channel.

Therefore we conclude that minimizing only a single autocorrelation provides the same channel shortening as obtained with a range of autocorrelations. Moreover, the choice of the lag is not fixed. Hence, we name our algorithm Any Lag Autocorrelation Minimization (ALAM) based channel shortening. In particular ALAM suggests any single lag from :

$$l = 1 \ to \ l = Lc$$

$$R_{yy}(l) = \sum_{j=0}^{L_c} c(j)c(j+l)$$

Will always be selected from

$$l = 1 \ to \ l = Lc$$

Since Cost function in Blind channel shortening is to be based upon the computation of minimal value of auto correlation. Therefore;

$$J_{ALAM} = Ryy(l) \quad , 0 < l \leq L_c \qquad (3.3.2)$$

It is interesting to note that whether we use one autocorrelation or a range of autocorrelations, the shortened channel and the TEQ designed are the same.and that is the fundamental concept to design ALAM.

Another observation which supports our claims in above paragraphs is that all of the autocorrelation based algorithms diverge once they reach the same optimal performance. This again supports our point of view that in these methods, minimization of the taps is without any regard to channel shortening by definition and hence carrying on updating the TEQ actually degrades the performance metric. We shall come back to this point again in the simulations chapter.

3.4. ALAM Algorithm

The stochastic gradient-descent algorithm to minimize J_{ALAM} is given below

$$w^{new} = w^{old} - \mu\nabla_w j_{ALAM} \qquad (3.3.3)$$

$$w^{new} = w^{old} - \mu\nabla_w \{E\,|y(n)y(n-l)|^2\}$$

$$w^{new} = w^{old} - \mu\nabla_w\,|y(n)y(n-l)|^2$$

It can further be simplified as

$$w^{k+1} = w^k - 2\mu \ y(n)y(n-1)\nabla_w(y(n)y(n-1))$$

$$w^{k+1} = w^k \ - 2\mu \ y(n)y(n-1)(y(n)r(n-l) + y(n-1)r(n)) \qquad (3.3.4)$$

Here μ is the step size and $0 < l \leq Lc.$ we have used instantaneous estimates of the autocorrelation which further reduces the computational complexity of the algorithm from SLAM. It is also more suitable for time varying channels as it updates the TEQ more frequently as compared to SAM which uses a averaging window to estimate the autocorrelations [9]. The last line of the equations is to make sure that the trivial solution of **w=0** is avoided. Selection of lag according to ALAM depends upon the variation in channel .If channel varying fast than auto correlation should be computed by selecting lag within lower range. But if channel varying slowly than lag should be selected with in higher range. Therefore as compare to pervious, ALAM provide the flexibility in lag selection in response to channel variation. Therefore adopting lag as in ALAM we can improve the effect of channel variation in channel shortening as was a critical problem in all viewed technique.

So By the virtue of ALAM, a best fitted lag can be approached, which can provide high SIR and to get significant improvement in out put bite rate. This technique as well as not only to defined a true sprit to achieve the minimal value of auto correlation to approached higher bite rate but also treated the drop down in bite rate as was the most severe problem in SAM, LHSAM etc. Moreover by considering the advantage of [22] as an optimum technique for fast varying channel, ALAM is mainly designed to update its parameter from sample to sample. Hence ALAM is best solution for optimum bite rate as well as to apply shortening in fast varying channel. Proper initialization of TEQ as well as precise weight normalization in true existent are badly needed to ensure appropriate convergence performance of ALAM, SAM, and SLAM cost function.

3.3. COMPLEXITY COMPARISONS

Complexity is usually difficult to measure precisely. Factors weighting the computational complexity are reusing computation, how much extra memory is available for storing previous computation rather than reusing them, whether the processing is performed in fixed point arithmetic or floating point arithmetic, and whether or not certain operation (i.e FFT) can be done in dedicated hardware.

Table 3 shows the (approximate) Computational complexity comparisons of SAM based technique and proposed approach (ALAM) for computing channel shortening.

Algorithm	Complexity (No. of multiplications)
SAM	$4N_{avg}L_w(L_c\text{-}v)$
SLAM, LHSAM, Khaled et al	$4N_{avg}L_w$
ALAM, Ishaq	$4L_w$

Table 1 Complexity Comparisons.

SLAM reduces complexity N times than SAM. Where as ALAM, reduces complexity N times than SLAM. Another advantage of ALAM is that its TEQ Updates tape weight instantly rather than nth times instant. Calculations shown in above tables are the comprehensible Operative difference between these algorithms. For Algorithm standard parameter like Filter Length, Step size, Input symbols, FFT size and CP length are same like other. Proposed method (ALAM) yields a little bit decrease in computational complexity rather than recently proposed Efficient Blind channel algorithm by Kemel Tape [9].

Chapter 4.

PERFORMANCE ASSESSMENT

4.1. Simulations

This section simulates the Low complexity blind adaptive channel shortening algorithm deprived in this dissertation and evaluates their asymptomatic performance to the performance of well-liked blind approached. For simulations, parameters were chosen to match the standard for ADSL downstream transmission and will use the CP of length v=32, the FFT of size 512, an equalizer of length Lw = 16, and the transmission channels will be the CSA test Loops [1~8] as being available at [10].They are real and energy will be in 200 consecutive taps .The 4-QAM signaling has been for each subcarrier respectively. These parameters were so chosen to compare the results with those of SAM in [3]. The signal to noise power was such that $\sigma_x^2 \|h\|^2 / \sigma_v^2 = 40\, dB$. This is a typical value of SNR in ADSL environments. A total of 20 symbols, each comprising of 544 samples are being employed here in ALAM. It is evident that proposed technique like "ALAM" is fast converging as it comprises instantaneous estimates of the autocorrelation. In continuation, 20 symbols are being simulated here to show the relative convergence in ALAM at diverse lags. For a point-to-point system alike with appropriate bit loading, the achievable bit rate for a fixed probability of error (typically 10^{-7} in DSL) is the performance metric [11]. The SNR gap Γ is given by

$$\Gamma = \Gamma_{gap} + \gamma_m - \gamma_c \tag{4.1.1}$$

The bit rate carrying each subcarrier is determined using noise margin γ_m = 6dB and the coding gain γ_c = 4.2 dB. The value of Γ_{gap} = 9.8 dB is used which corresponds to a probability of error 10^{-7} and the QAM modulation used across the subcarriers. The bit rate on each subcarrier i is calculated as under [11]

$$b = \sum_{i=1}^{N/2} \log_2[1 + \frac{SNR}{\Gamma}]$$

The noise definition in SNR_i in Equation (9) includes the channel noise as well as the distortion due to ICI and ISI caused by the energy of the channel outside the v +1 length. Finally, the bit rate is computed with the formula

$$rate = (\sum_{i=1}^{N/2} b)\frac{Fs}{N+V} \tag{4.1.2}$$

This metric is used to assess the performance of the TEQ algorithms. Here F_s = 2.208 MHz is the sampling frequency. We will also explore the plot of SSNR of ALAM algorithm. The step sizes for ALAM remains same throughout all the respective lags. This process has been performed for such to show the relative convergence in ALAM. The initialization was single center spike. The performance of ALAM algorithm has also been compared with that of MSSNR method [2].and results are displayed here to get confirmation in performance assessment. Moreover, the capacity of the system will be:

$$C = \sum_{n=1}^{N} \log_2(1 + \frac{SNR}{\Gamma}) \tag{4.1.3}$$

Channel capacity menially depends upon the SNR values and to corresponding bite rate. Therefore bite rate is an important parameter in performance

4.2. ALAM Results

Figure 5 shows the original CSA 1 Loop channel and shortened channel with ALAM using lag=1. Figure 6 shows the original CSA 1 Loop channel and shortened channel with MSSNR method. Both algorithms shorten the channel but the shape of the channel is different inside the window. Figure 7 shows the TEQ designed with ALAM using lag=1 while Figure 8 shows the TEQ designed with MSSNR method. Again the shapes of the two TEQ are different.

In Figure 9 the top to bottom plots show the SSNR, Achievable bit rate, ALAM cost, and SAM cost, respectively versus the iteration number. ALAM using lag=1 first achieves bit rates close to that of MSSNR but then diverges. We also show that by minimizing ALAM cost, we essentially also minimize SAM cost. Therefore, ALAM and SAM behave similarly although ALAM complexity is too, low.

Figure 10 shows one sided autocorrelation of the CSA 1 channel and that of the shortened channel with ALAM using lag=1. The autocorrelation has been shortened with ALAM using a single autocorrelation as compared to SAM using a range of autocorrelations. This might be a coincidence, but we have demonstrated that at least for ADSL channels, ALAM behaves identical to SAM.

Figure 10 shows one sided autocorrelation of the CSA 1 channel and that of the shortened channel with ALAM using lag=1. The autocorrelation has been shortened with ALAM using a single autocorrelation as compared to SAM using a range of autocorrelations. Figure 11 shows one sided autocorrelation of the CSA 1 channel and that of the shortened channel with MSSNR method. It shows the success of both the algorithms in shortening the channel. We recall that a shortened channel will definitely have a short autocorrelation. But in last chapter we have elaborated that the reverse is not true.

Similar comments apply to Figures 12-15. In these figures, ALAM used lag=33. As the channel being simulated is same CSA Loop 1, so there is no need to repeat the MSSNR results. Similar comments also apply to Figures 12-15. In these figures, ALAM used lag=400. Again as the channel being simulated is same CSA Loop 1, so there is no need to repeat the MSSNR results.

It should be noted that the ALAM TEQ, its shortened channel, and the shortened autocorrelation are shown for the iteration when it achieves its preeminent bit rate. The point to demonstrate is that ALAM using different lag values essentially does identical channel shortening. In the relevant figures, we can see that the TEQ designed, the shortened channels and the shortened autocorrelations are identical.

In Figure 16, top to bottom plots show the Energy inside, Energy outside, and their ration (SSNR) in a window of CP length, respectively versus the iteration number. ALAM had lag=400. We see that ALAM, and as a matter of fact all autocorrelation based channel shortening algorithms during adaptation to minimize the energy inside as well outside. It is a mere chance that the SSNR increases at the some point. After that point, the decreasing trend in inside and outside energy continues resulting to drop in the achievable data rate metric. Compare this Figure with Figure 15. Note that in Figure 15, SSNR has been shown in dB scale.

Figure 8 and Figure 22 demonstrate the same in another way. In Figure 21, the original CSA 1 Loop channel and shortened channel with ALAM using lag=400 is shown. This time ALAM is initialized with the MSSNR TEQ as to be compared with the Figure 22. ALAM was effectively provided a shortened channel at the start of updating (which is shown in Figure 22) and it has just minimized the norm of the already shortened channel.

Likewise, we have also simulated ALAM devising some other lags= 100,200,500. Other CSA Loop channels 2-8 were also simulated. The results (not shown here to avoid repetition) are similar and back our claims.

Conclusions will be demonstrated in the next chapter.

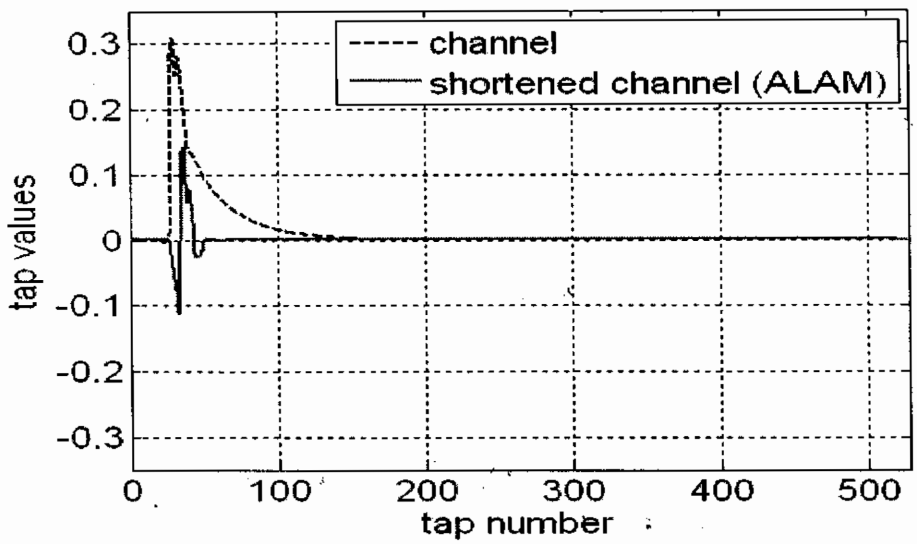

Figure 5.Original CSA 1 Loop channel and shortened channel with ALAM using lag=1.

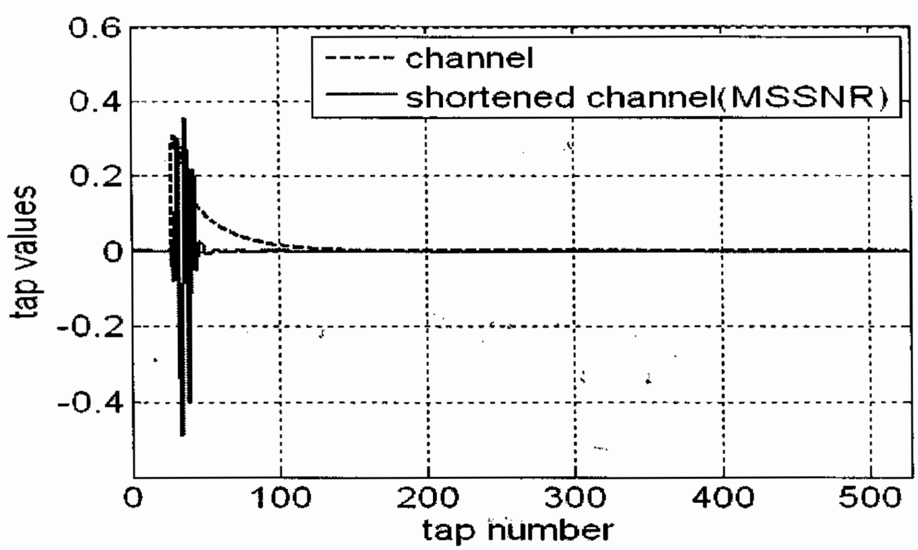

Figure 6. Original CSA 1 Loop channel and MSSNR based Shortened channel.

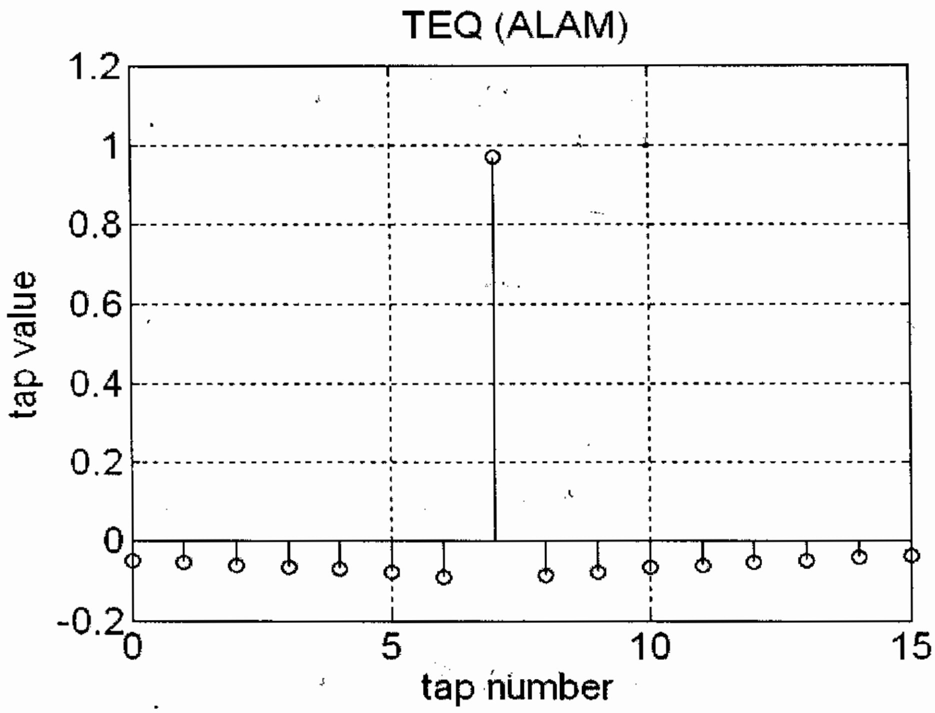

Figure 7. TEQ designed with ALAM using lag=1.

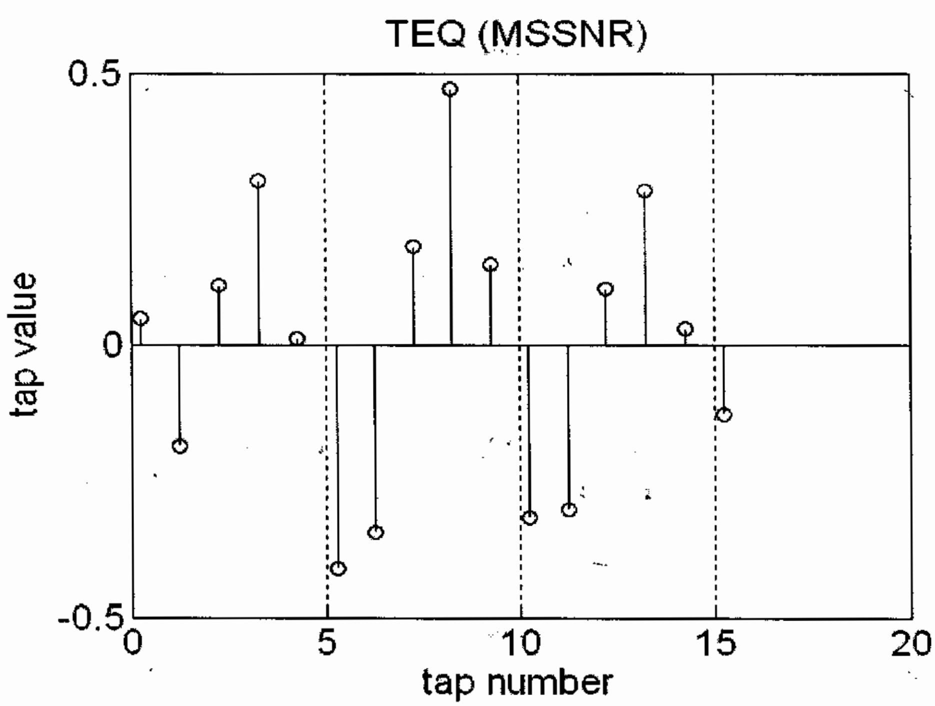

Figure 8. TEQ designed with MSSNR method.

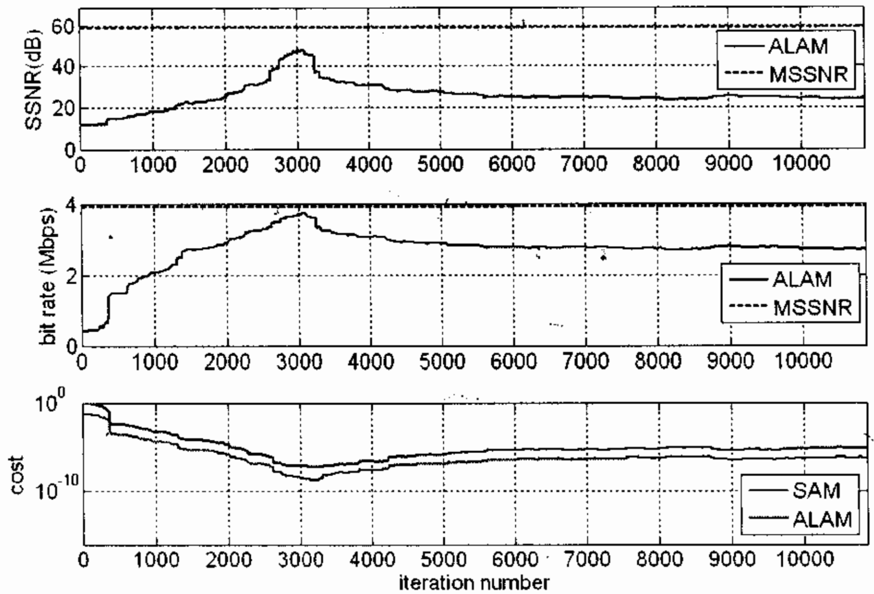

Figure 9. SSNR, Achievable bit rate, ALAM cost

Figure 4.5 Top to bottom plots show the SSNR, Achievable bit rate, ALAM cost, and SAM cost, respectively versus the iteration number. ALAM using lag=1 first achieves bit rates close to that of MSSNR but then diverges. We also show that by minimizing ALAM cost, we essentially also minimize SAM cost. Therefore, ALAM and SAM behave similarly although ALAM complexity is too, low.

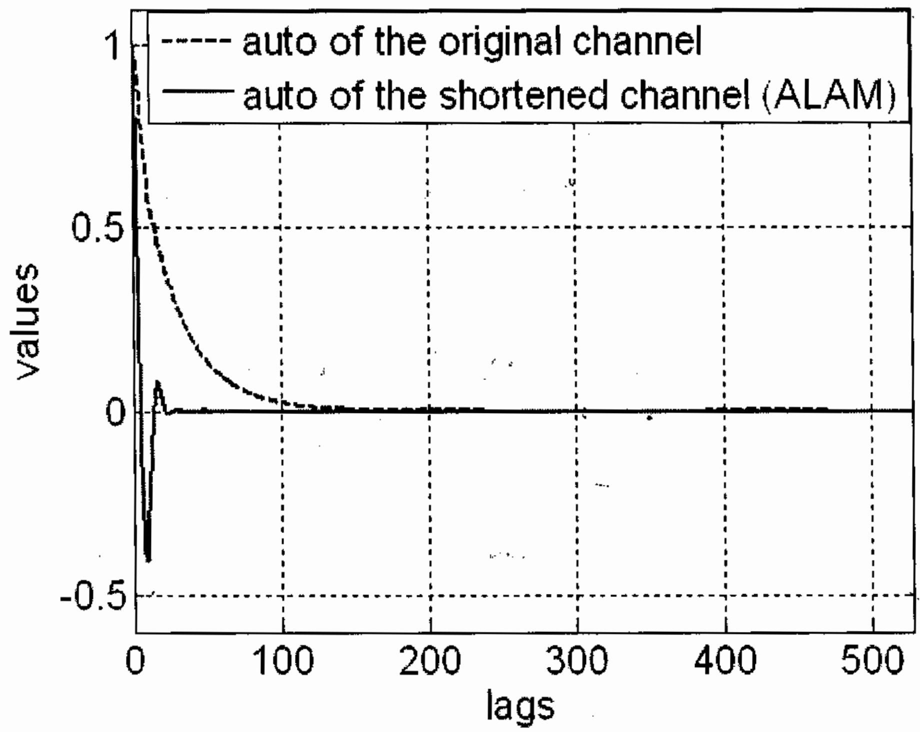

Figure 10. Autocorrelation of the CSA Loop 1 and shortened channel

Figure 10 One sided autocorrelation of the CSA 1 channel and that of the shortened channel with ALAM using lag=1. The autocorrelation has been shortened with ALAM using a single autocorrelation as compared to SAM.

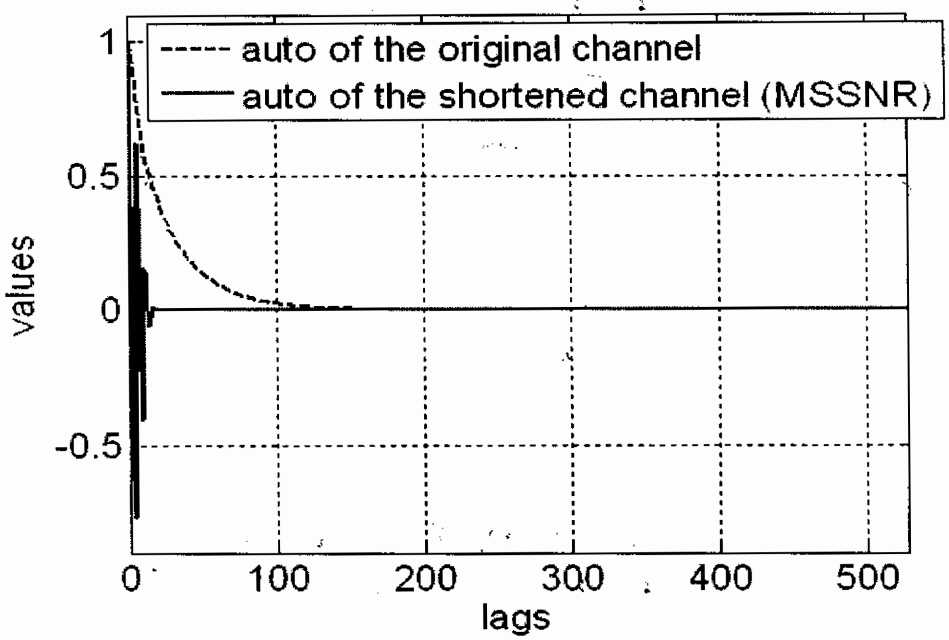

Figure 11. Autocorrelation of CSA Loop1 and MSSNR shortened channel.

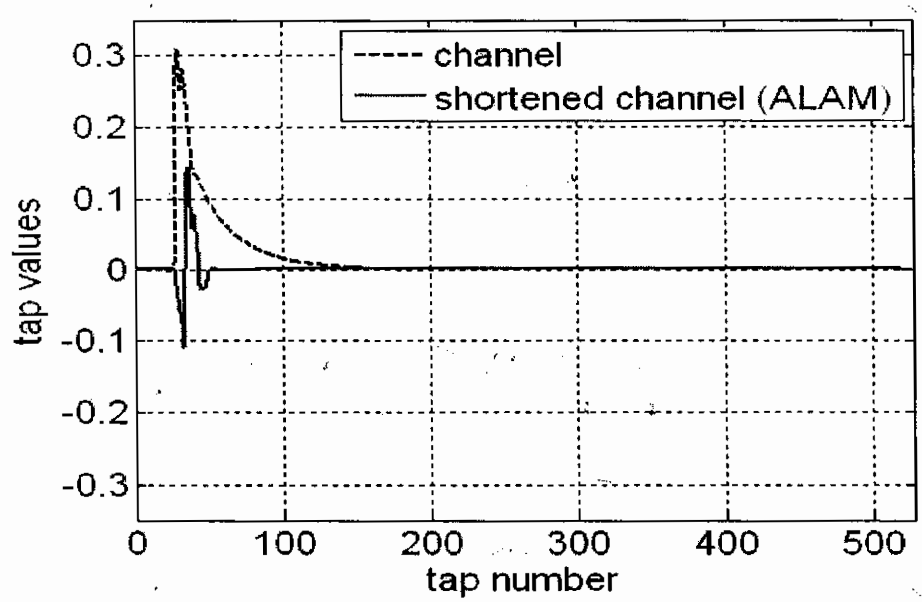

Figure 12. Original CSA 1 Loop and shortened channel with ALAM using lag=33.

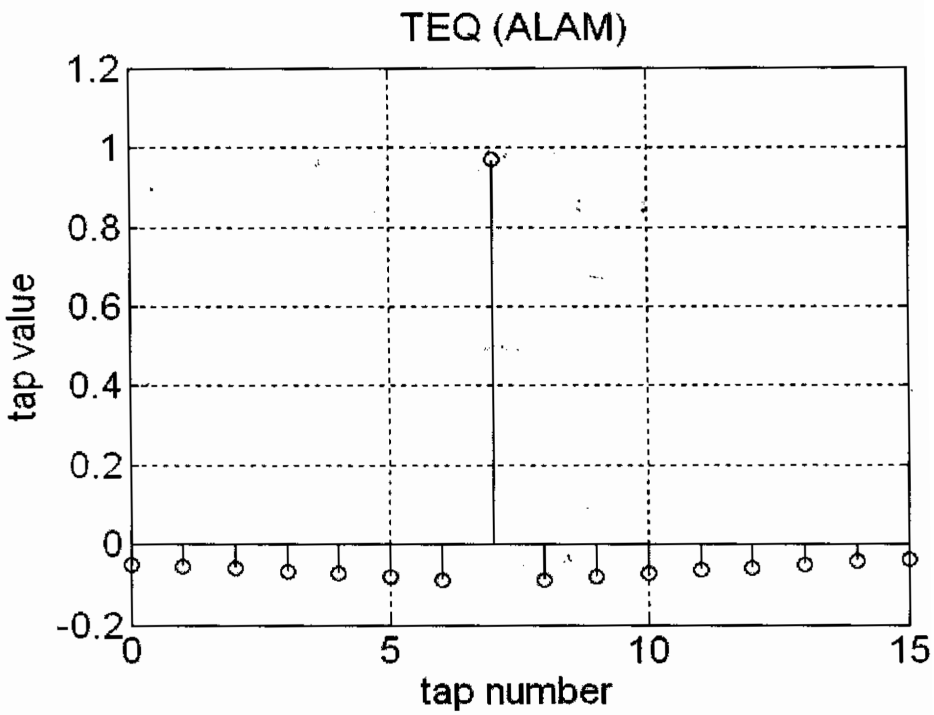

Figure 13. TEQ designed with ALAM using lag=33.

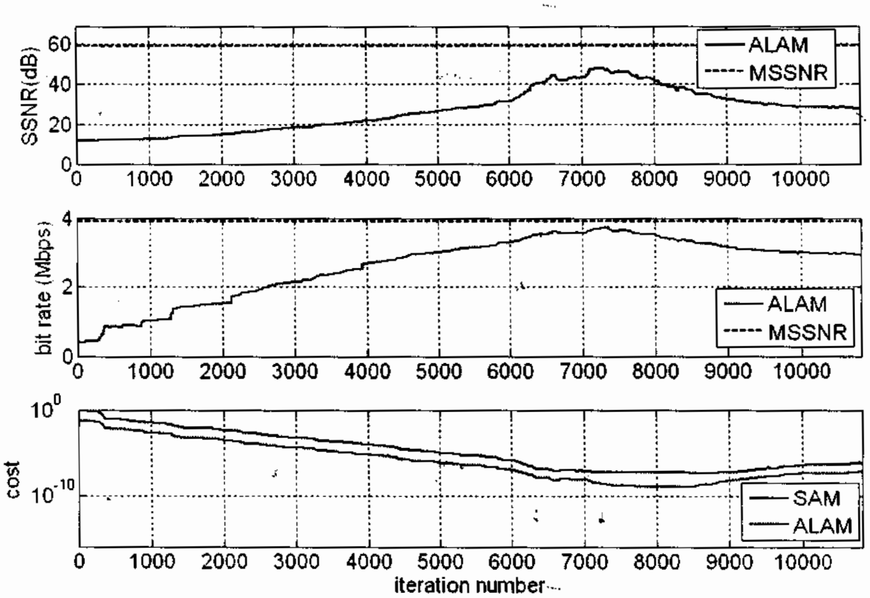

Figure 14. SSNR, Achievable bit rate, ALAM cost

Top to bottom plots show the SSNR, Achievable bit rate, ALAM cost, and SAM cost, respectively versus the iteration number. ALAM using lag=33 first achieves bit rates close to that of MSSNR but then diverges. We also show that by minimizing ALAM cost, we essentially also minimize SAM cost. Therefore, ALAM and SAM behave similarly although ALAM complexity is too, low.

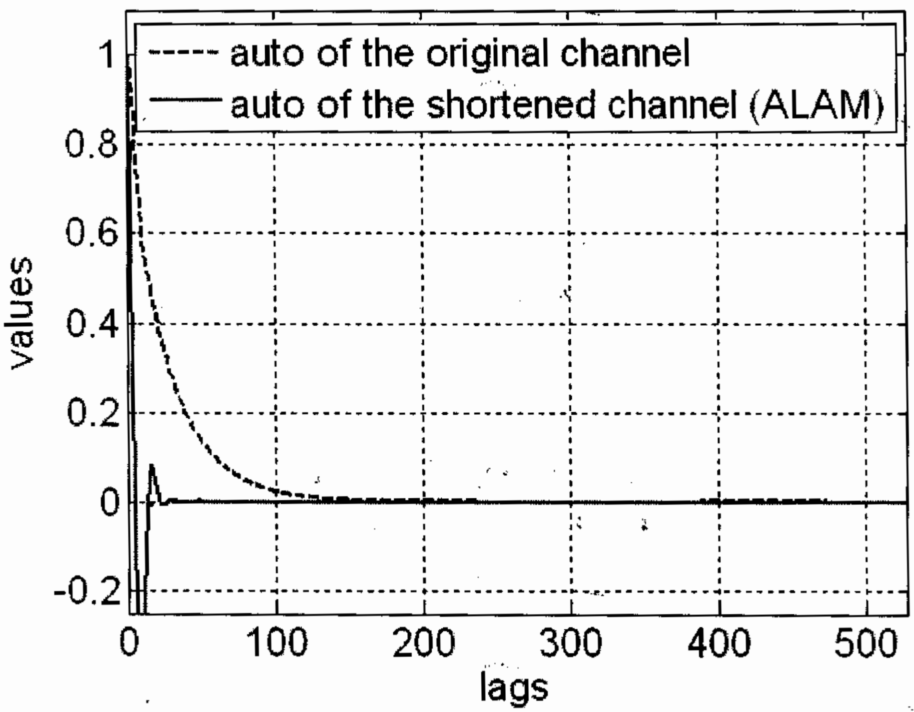

Figure 15. Autocorrelation of the CSA Loop1 and shortened channel

Figure 15 represents the one sided autocorrelation of the CSA 1 channel and that of the shortened channel with ALAM using lag=33. The autocorrelation has been shortened with ALAM using a single autocorrelation as compared to SAM.

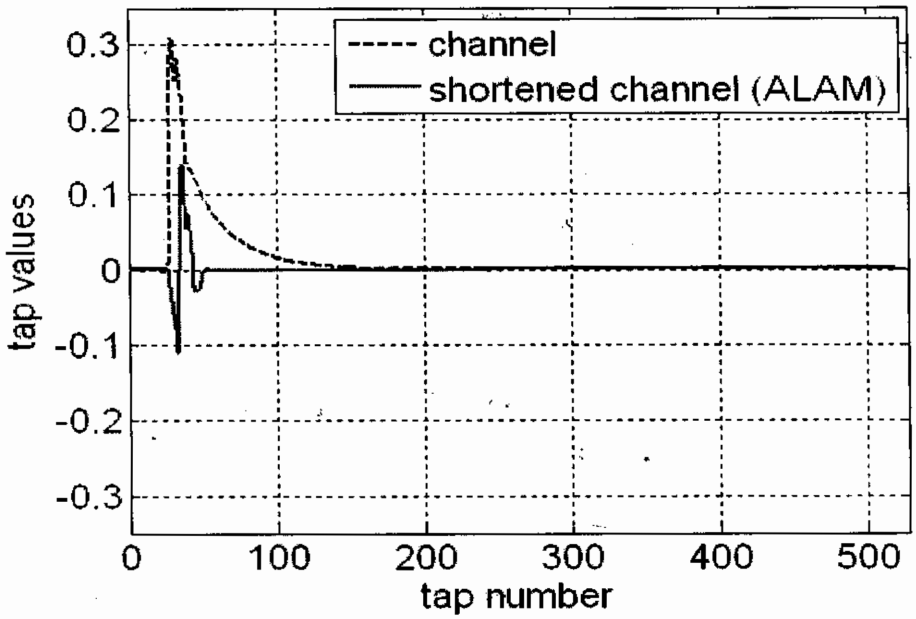

Figure 16. CSA Loop 1and shortened channel with ALAM using lag=400.

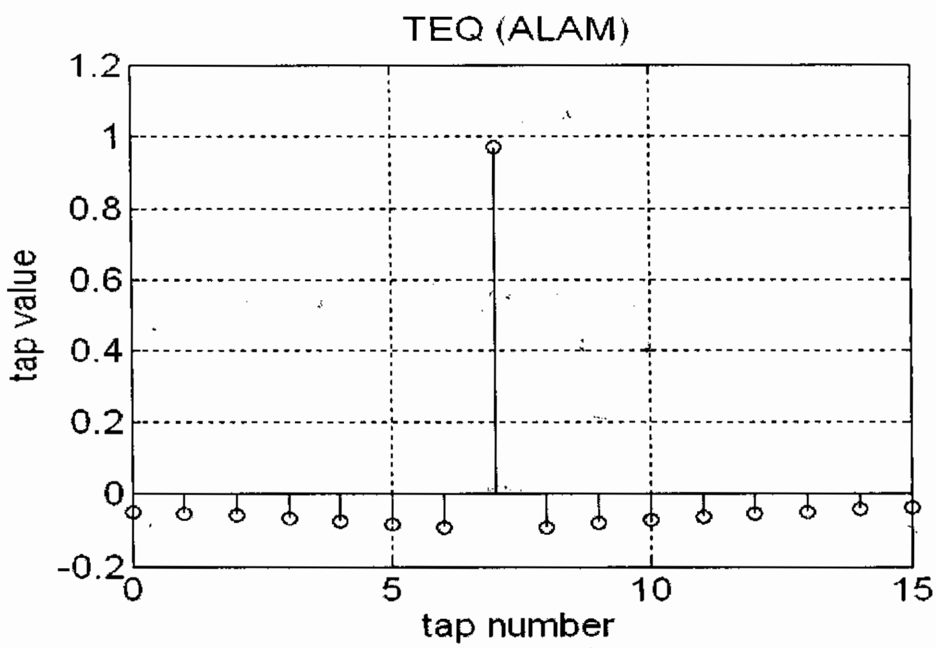

Figure 17: TEQ designed with ALAM using lag=400.

In previous, we notice that all the three TEQs designed using lag=1, 33, 400 are same.

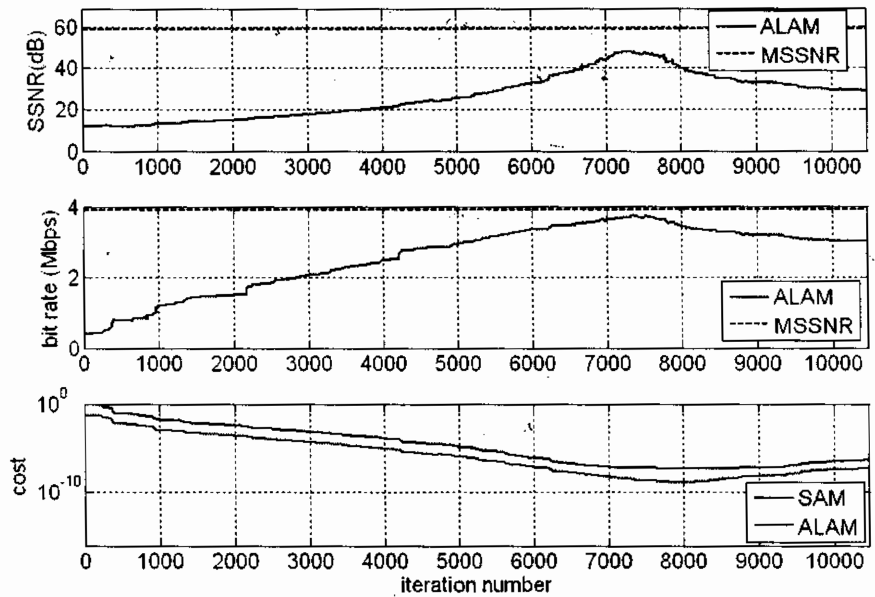

Figure 18.SSNR /Achieve able Bite rate/ALAM Cost

Figure 18Top to bottom plots show the SSNR, Achievable bit rate, ALAM cost, and SAM cost, respectively versus the iteration number. ALAM using lag=400 first achieves bit rates close to that of MSSNR but then diverges. We also show that by minimizing ALAM cost, we essentially also minimize SAM cost Therefore, ALAM and SAM behave similarly although ALAM complexity is too low.

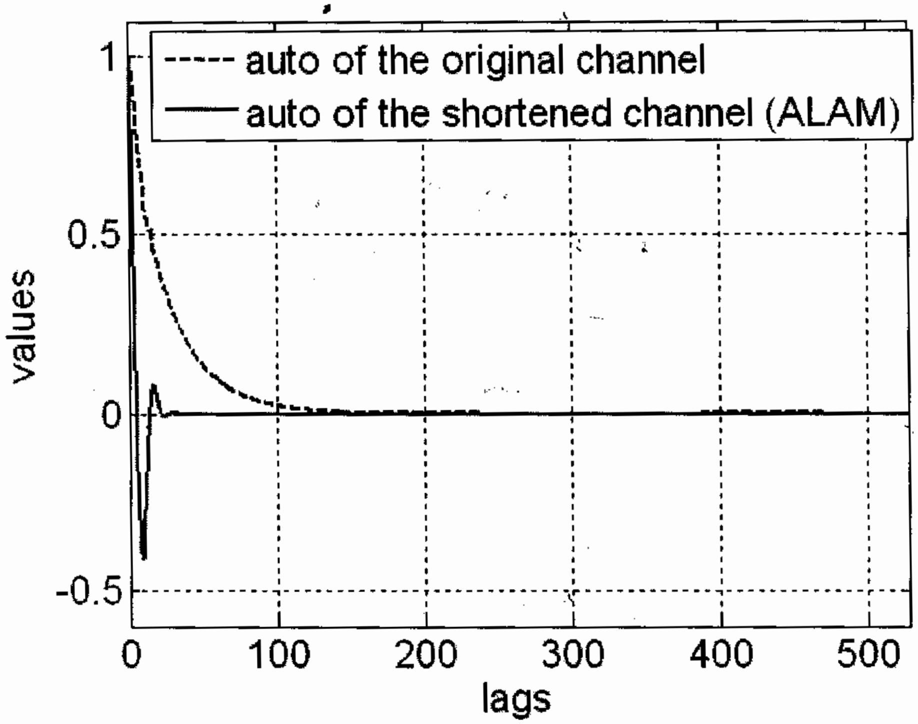

Figure 19. Autocorrelation of CSA 1 and ALAM shortened channel using lag=400

Figure 19 One sided autocorrelation of the CSA 1 channel and that of the shortened channel with ALAM using lag=400. The autocorrelation has been shortened with ALAM using a single autocorrelation as compared to SAM.

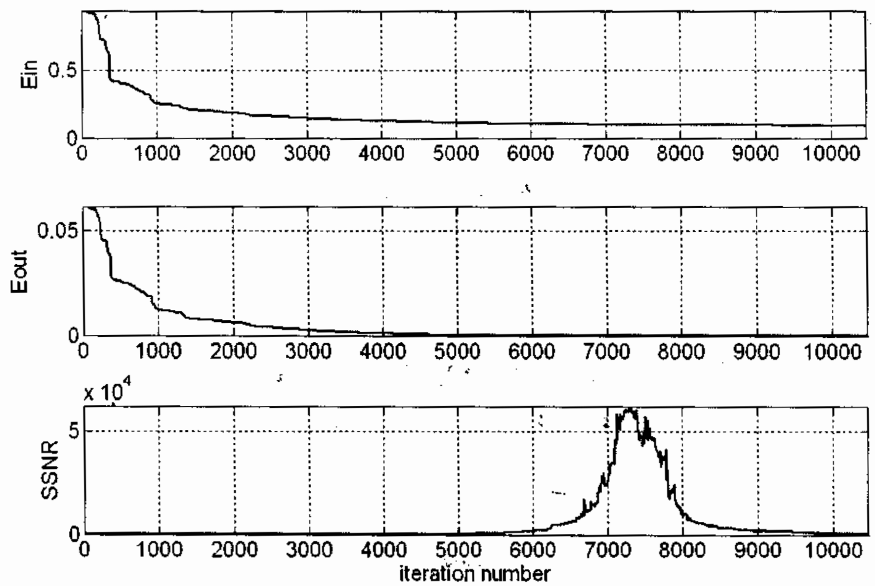

Figure 20. Energy inside, Energy outside, and their ration within CP length

Figure 20 Top to bottom plots show the Energy inside, Energy outside, and their ration (SSNR) in a window of CP length, respectively versus the iteration number. ALAM had lag=400. We see that ALAM, and as a matter of fact all autocorrelation based channel shortening algorithms during adaptation minimize the energy inside as well outside. It is a mere chance that the SSNR increases at some point. After that point, the decrease in inside and outside energy keep on decreasing resulting in the drop in the data rate metric. Compare this Figure with Figure 4.11. Note that in Figure 4.11, SSNR has been shown in dB scale.

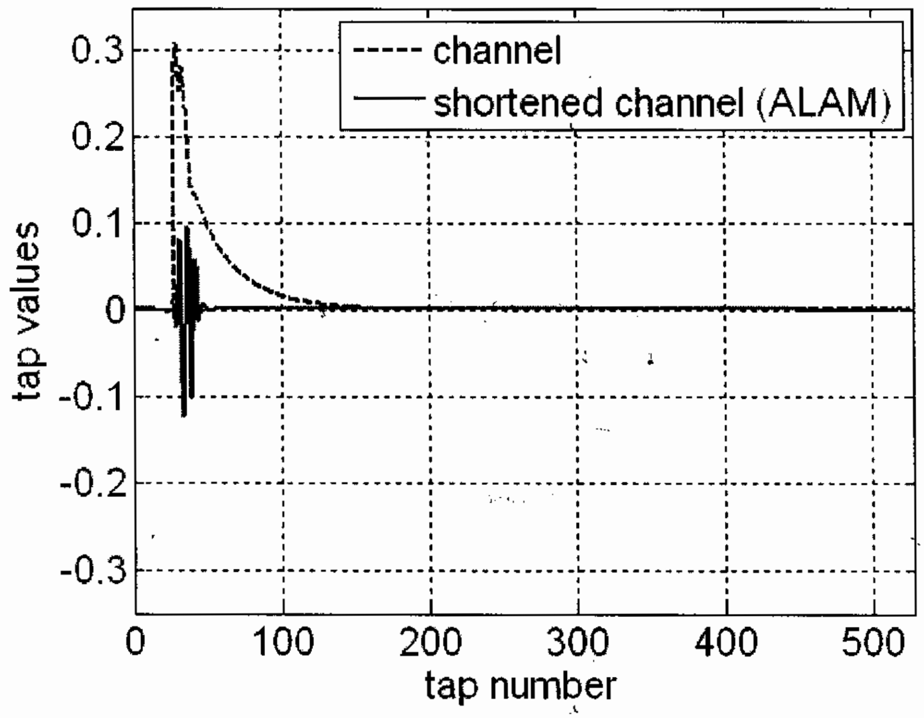

Figure 21. Original CSA 1 Loop and shortened channel with ALAM using lag=400.

Figure 21 Original CSA 1 Loop channel and shortened channel with ALAM using lag=400. ALAM is initialized with the MSSNR TEQ. Compare with the next Figure. ALAM was provided a shortened channel at the start of updating (shown in next figure) and it has just minimized its norm.

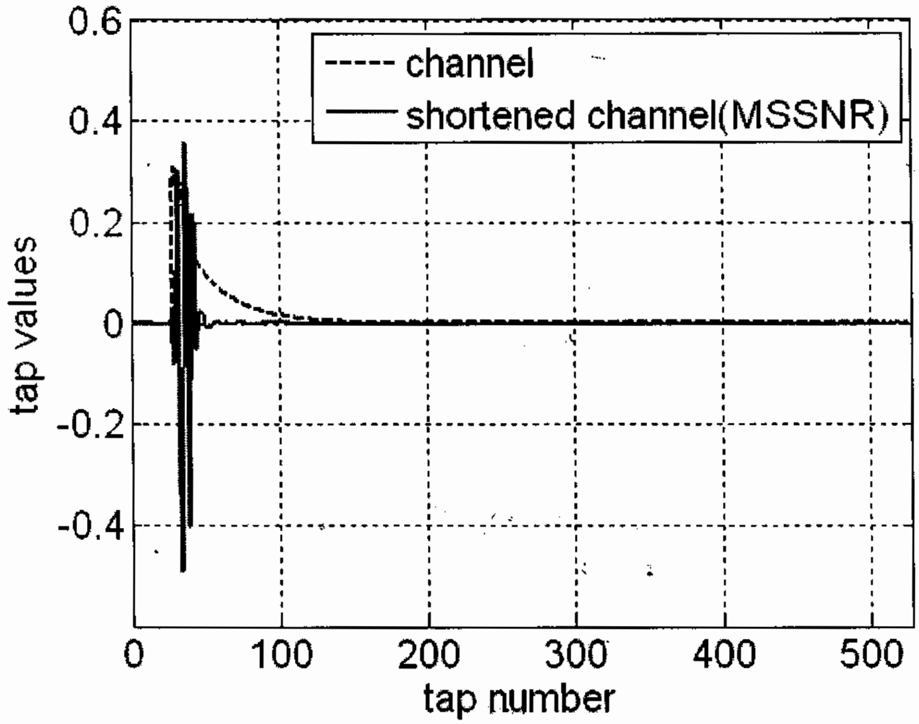

Figure 22. Original CSA 1 Loop channel and shortened channel with MSSNR method.

Chapter 5.

CONCLUSIONS AND FUTURE WORK

5.1. Conclusions and Future work

SAM is based on the fact that; if a short filter introduces short autocorrelation in its output the reverse might be true [12].We have demonstrated that the reverse is not true. The CSA Loop channels have been shortened with "ALAM" using a single autocorrelation as compared to SAM using a range of autocorrelations. We have demonstrated that at least for ADSL channels, "ALAM" behaves identical to "SAM" and present an extremely low complexity substitute.

ALAM serves as an insight into the autocorrelation based channel shortening algorithms. The details of chapter 3 has been shown clearly in different type of window control, we have in order to perform channel shortening. It shows that using a range of autocorrelations is unintelligent and just overkill.

Unfortunately, the window placement control is not guided by the nature of the channel. Different lag options for ALAM simply put different amount of minimization on different portions of the channel. One thing is common, they start minimizing the norm of the channel without any regard to channel shortening by definition. Channel shortening requires that we minimize the channel but avoid a contagious window of taps from minimization. This is the reason these algorithms diverge once achieving their best performance. The bit rate metric penalizes the SSNR of the shortened channel. The only way divergence of ALAM (and, as a matter of fact, of SAM) for shortening the ADSL channels can be stopped is by freezing the adaptation (by decreasing the step size to zero) when the best performance has been achieved. One heuristic was proposed by Nawaz et al in [4]. Other solutions might be suggested.

However proposed model has proven improvement in Signal to Interference Ratio (SIR) like SAM and SLAM. Selection of lag proceeding to ALAM depends upon the variation in channel .If channel varying fast than auto correlation computation must be carried by selecting lag within lower range. But if channel varying slowly than lag should be selected with in higher range. Therefore as compare to pervious, ALAM also provide the flexibility in lag selection in response to channel variation.

5.2. Future Work

For some other channels, ALAM might not perform identical to SAM. One future direction would be to compare SAM and ALAM algorithms for other channels, like DAB and WLAN channels. Our intuition is that ALAM is going to succeed and will also not diverge if the channel has high energy concentrated in the centre and a careful selection of lag has been made; on the other hand, SAM would be blindly minimizing every type and is not likely to shorten the channel. We have given guidelines in chapter 3 on how choice of different lags affects ALAM operation. This guideline gave a detailed insight the channel impulse response and with relation to auto correlation.

Although apart from degradation in bite rate, Technique can be best fitted to some extent like in wireless channel being tuned variation with varying delay spread. Although designed work in ALAM, immune to contribute any proficiency in performance degradation like Computational complexity, optimum bite rate, and fast convergences etc but it extends all those feature in to a generalized method while retains it's all performance matrices in true spirit.

References

[1] R. Nawaz and J. A. Chambers, "Low complexity blind adaptive channel shortening and equalization for Multi-Carrier systems", PhD thesis Cardiff University, UK, 2006.

[2] P. J. W. Melsa, R. C. Younce, and C. E. Rohrs. "Impulse Response Shortening For Discrete Multitone Transceivers". IEEE Trans. on Comm., volume.44, pp.1662 ~1672, December .1996.

[3] J.Balakrishnan, R.K.Martin and C.R. Johnson,Jr,"Blind, Adaptive Channel Shortening by Sum Squared Auto Correlation Minimization (SAM),"IEEE Trans.Signal Process. VOL.51, no.12, PP.3086-3093, 2003.

[4] R.Nawaz and J.A Chamber, "A novel Single Lag Autocorrelation Minimization Algorithm for Blind adaptive channels shortening", "ICASSP", VOL. 3, no.25, PP.885-888, 2005.

[5] M. Girara and J, A Chambers, "A blind Lag Hopping Adaptive Channel Shortening Algorithm Based Upon Squared Auto Correlation Minimization (LHSAM)," In Proc. Int. Conf. on Acoustic, Speech, and Signal Processing (ICASSP), pp. 3565~3572, 2008.

[6] Khaled and J. A. Chambers, A generalized blind lag hopping adaptive channel shortening algorithm based upon squared autocorrelation minimization, PhD thesis Cardiff University 2008.

[7] C. Yin and G. Yue. "Optimal Impulse Response Shortening for Discrete multi tone Transceivers". , Electronics Letters, vol.34, pp.35~36, January. 1998.

[8] S. M. Kay, Intuitive probability and random processes using Matlab, Springer 2006.

[9] Ishaq Gul Muhammad, Esam Abdel Raheem, Kemel Tepe,"Efficient Blind Adaptive Channel Shortening Algorithm for Multicarrier Modulation System", In IEEE International Symposium on signal processing and Information Technology(ISSPIT),2009.

[10] RK Martin "Matlab Code for Paper by R.K Martin, Available: Http: // bard .ece.Cornell.edu/ mat lab /martin/index.html"

[11] G. Arslan, B. L. Evans, and S. Kiaei. Equalization for Discrete Multitone Receivers to Maximize Bit Rate,"IEEE Trans. Signal Processing, vol.49, pp.3123~3135, Dec. 2001.

[12] R. K. Martin Signal Processing magazine "Adaptive Equalization: Transitioning from Single Carrier to Multicarrier System" vol.22, No.6, pp.108[1] R. Nawaz and J. A. Chambers, "Low complexity blind adaptive channel shortening and equalization for Multi-Carrier systems", PhD thesis Cardiff University, UK, 2006.

[13] R. K. Martin Signal Processing magazine "Adaptive Equalization: Transitioning from Single Carrier to Multicarrier System~122, Nov.2005.

[14] R. Schur and J. Speidel. An Efficient Equalization Method to Minimize Delay Spread in OFDM/DMT Systems. In Proc. IEEE Int. Conf. on Comm., volume 5, pages 1481{1485, Helsinki, Finland, June 2001.

[15] M. de Courville, P. Duhamel, P. Madec, and J. Palicot. Blind equalization of OFDM systems based on the minimization of a quadratic criterion. In Proc.EEE Int. Conf. on Comm., pages 1318{1321, Dallas, TX, June 1996.

[16] N.Al-Dhahir Evans, and J.M.Cioffi,"Optimum Finite Length Equalization for Multicarrier Transmission,"IEEE Trans on Comm., VOL 44, NO.1, pp.56-64, jan.1996.

[17] J.S.Chow, J.M.Cioffi and J.A.C Bingham; Equalizer Training Algorithm for Multicarrier Modulation System, "IN Proc. IEEE Int.Conf. On Comm, May 1993.pp.761-765.

[18] J. S. Chow, J. M. Cioffi, and J. A. C. Bingham, "Equalizer training Algorithms for Multicarrier modulation systems," in Proc. IEEE Int.Conf. On Common, pp. 761–765. Geneva, Switzerland, May 1993.

[19] M. Milosevic's, "Maximizing Data Rate of Discrete Multitone Systems Using Time Domain Equalization Design," Ph. D. Thesis, the University of Texas at Austin 2003.

[20] J. F. V. Kerchief and P. Spruyt, "Adapted optimization criterion for FDM-based DMT-ADSL equalization," in Proc. IEEE Int. Conf.Commun., pp. 1328–1334. June 1996.

[21] M. Nae and A. Gatherer. Time-Domain Equalizer Training for ADSL. InProc.IEEEInt. Conf. on Comm., volume 2, pages 1085~1089, Montreal, Canada, June1997.

[22] R. K. Martin, J. Balakrishnan, W. A. Sethares, and C. R. Johnson, Jr.A Blind, Adaptive TEQ for Multicarrier Systems. IEEE Signal Processing Letters, 9(11):341~343, November 2002.

[23] J.M Walsh, RK Martin, CR Johnson: Convergence and performance issues for Auto Correlation based channel shortening,"Proc 40th Aalsilomar Conf signal, system and computer, 2006.

[24] B. L. Evans, "Equalizer Design to Maximize Bit Rate In ADSL Transceivers," Lecture on ADSL Transceivers Dept. of Electrical and Comp. Eng. The University of Texas at Austin [Online July 16, 2006] Available http://www.ece.utexas.edu/~be vans/projects/adsl/index.html

[25] P. S. Chow, J. M. Cioffi, and J. A. C. Bingham, "A practical discrete Multitone transceiver loading algorithm for data transmission over spectrally shaped channels," IEEE Trans. on Common., vol. 43, no. 234, pp. 773–775, Feb./Mar./Apr. 1995

[26] D. D. Falconer and F. R. Magee, "Adaptive channel memory truncation for maximum likelihood sequence estimation," Bell Sys. TechJournal, pp. 1541–1562, Nov. 1973.106

[27] Medvedev and V. Tarokh, "A channel-shortening multiuser detector for DS CDMA systems," in Proceeding of the 53rd Veh. Tech. Conf., pp. 1834–1838. vol. 3, Rhodes, Greece, May 2001.

[28] S. I. Husain and J. Choi, "Single correlation based UWB receiver through channel shortening equalizer," in 2005 Asia-Pacific Conf. on Common., pp. 610–614. Perth, Western Australia, Oct. 2005.

[29] M. Kallinger and A. Martins, "Room impulse response shortening by channel shortening concepts," in Proc. IEEE Asilomar Conf. on Signals, Systems and Comp., pp. 898–902. Pacific Grove, CA, Nov.2005.

[30] K. V. Acker, G. Leus, M. Moonen, O. van de Wiel, and T. Pollet, "Per tone equalization for DMT-based systems," IEEE Trans. Com-Mun. vol. 49, no. 1, pp. 109–119, Jan. 2001.

[31] M. Milosevic's, "Maximizing Data Rate of Discrete Multitone Systems Using Time Domain Equalization Design," Ph. D. Thesis, the University of Texas at Austin 2003.

[32] M. G. Troilus and S. Sesia, "A spectrally flat time domain equalizer for rate improvement of multicarrier systems," in Proc. IEEE Int. Conf.Commun., pp. 1803–1807. May 2002.

[33] J. F. V. Kerchove and P. Spruyt, "Adapted optimization criterion for FDM-based DMT-ADSL equalization," in Proc. IEEE Int. Conf.Commun., pp. 1328–1334. June 1996.

[34] A. V. Oppenheim, R. W. Schafer, and J. R. Buck, Discrete-Time Signal Processing,2nd edition. Prentice-Hall, 1989.

[35] J. A. C. Bingham, "Multicarrier modulation for data transmission an idea whose time has come," IEEE Common. Magazine, vol. 28, no.5, pp. 5–14, May 1990.

[36] T. Starr, J. M. Cioffi , and P. T. Silverman, Understanding Digital Subscriber Line Technology. Englewood Cliffs NJ: Prentice-Hall, 1999.

[37] M. V. Bladel and M. Moeneclaey, "Time-domain equalization for multicarrier communication," in Proc. IEEE Global Telecomm. Conf., pp. 167–171. Nov. 1995