NARMA Equalizer for Nonlinear ITU channels with Modified Feedback

Murchana Boruah, Aradhana Misra, Manash Pratim Sarma, Kandarpa Kumar Sarma and Nikos Mastorakis

***Abstract*—** A Rayleigh fading channel is nonlinear in nature. Therefore channel modeling techniques can be more accurately represented by nonlinear stochastic models. In this paper, we consider a block faded Single Input Single Output (SISO) system. A Nonlinear Autoregressive Moving Average (NARMA) filter is designed to equalize the received signal. A double feedback equalizer structure is employed to make the training of the system more detailed. The effectiveness of this method is validated by simulations. Bit Error Rate (BER) vs. Signal to Noise Ratio (SNR) curve is analyzed for different channel conditions specified by Third Generation Partnership Project (3GPP) standards and International Telecommunication Union (ITU) specifications. Performance is evaluated for different training overheads and a tradeoff between spectral efficiency and performance is obtained. The proposed equalizer structure gives better performance in severe fading environments as well as in a low percentage of training symbols. The entropy is also increased when compared to a single feedback NARMA receiver structure.

***Keywords*— Nonlinear, Stochastic, NARMA, Equalizer, Channels, ITU**

I. INTRODUCTION

Fading is a widely observed phenomena in wireless communications. Fading occurs when a signal propagating through a wireless medium undergo change in its phase and amplitude due to refraction, diffraction and scattering from different objects present in the medium [1]. Apart from this, another factor, Doppler frequency, which is a direct cause of the relative motion between the transmitter and the receiver also plays a major role in determining the extent of fading [1]. Several methods have been investigated for proper and efficient recovery of transmitted symbols at the receiver end. This has necessitated modeling of fading channels that would resemble real time channels. In long range transmissions when the channel is changing continuously, or the receiver is moving at a high speed, severe fading occurs. In such situations, the nonlinear components of the channel cannot be ignored. The nonlinearity of a channel must thus be taken into account for designing channel models. The conventional method of using Jakes's Doppler spectrum to represent the channel proved inefficient compared to stochastic channel models as shown in [2]. Stochastic channel models such as Autoregressive (AR), Moving Average (MA), Autoregressive Moving Average (ARMA) models are highly used as shown in [2] to [4] for its random behaviour similar to a Rayleigh channel. To give a more detailed representation of the channel, nonlinearities have been introduced in these models to evaluate the signal in severe fading environments. Introducing nonlinearities in the system, for example, the nonlinear AR (NAR) or the nonlinear ARMA (NARMA) make the system complex and requires a large number of training symbols [5], [8]. This can be also be accomplished by Artificial Neural Network (ANN) models but it requires a huge computation time during training. Some works involving channel modeling using nonlinear approaches are [9] to [11]. In this work, we simplify the nonlinear system by using a Gaussian function as in [5] which reduces complexity and gives good performance.

Different channel equalization schemes are used to mitigate the harmful effects of fading which is caused by the Inter Symbol Interference (ISI) from adjacent symbols due to multipath propagation. Conventional equalization schemes like Least Square (LS), Minimum Mean Square Error (MMSE) are linear based methods [12], [13] and does not give better performance during severe fading conditions. On the other hand, traditional nonlinear equalizers like Decision Feedback Equalizer (DFE) give comparatively better performance than the linear methods [14], [15], however it suffers from a noise propagation problem [1] that may degrade the performance of the system. Equalization has also been achieved by means of both linear and nonlinear regression [16] to [19]. In case of AR model, complications arise in deciding the model order [19].

In this paper, we propose a modified structure of the NARMA equalizer presented in our previous work [5] with a double feedback structure. The double feedback structure enables a detailed learning of the channel characteristics by the NARMA equalizer. Section II elaborates the related theoretical considerations, Section III describes the proposed model, Section IV discusses the experimental details and results and Section V concludes.

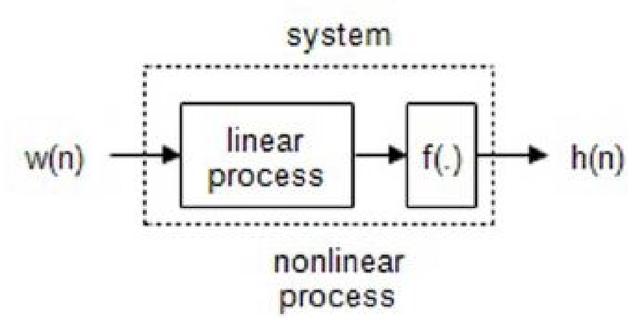

Figure 1: NARMA model

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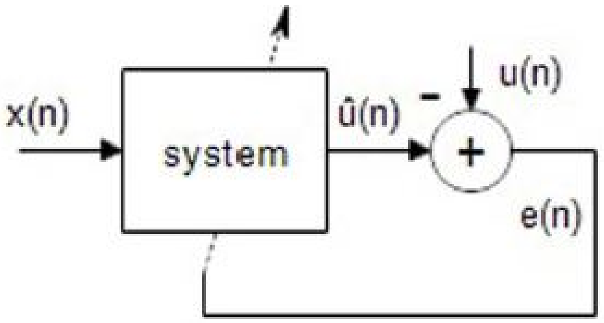

Figure 2: Adaptive Filter

II. RELATED CONSIDERATIONS

This section briefly discuss the theoretical concepts relevant to the work and are attributed to nonlinear Rayleigh channel modeling as well as Prony's method to determine filter coefficients. It is divided into three subsections. Subsection 2.1 describes the NARMA channel model and its mathematical expression, Subsection 2.2 discusses about adaptive filtering and Subsection 2.3 discuss about Prony's method for obtaining ARMA model coefficients.

A. NARMA Model

The nonlinearities of a stochastic system can be represented by means of a NARMA model whose expression is given in eq. 1.

$$h(n) = \sum_{i=0}^{i=p} a(i)\,\omega(n-i) + \sum_{j=0}^{j=q} b(j)\,h(n-j) + \sum_{i=0}^{i=p}\sum_{j=0}^{j=q} a(i,j)\,\omega(n-i)\omega(n-j) + \sum_{i=0}^{i=p}\sum_{j=0}^{j=q} b(i,j)\,h(n-i)h(n-j) + \sum_{i=0}^{i=p}\sum_{j=0}^{j=q} c(i,j)\,\omega(n-i)h(n-j) \ldots\ldots (1)$$

In eq. 1, $h(n)$ is the input process and $w(n)$ is the gaussian noise process as shown in Figure 1. Here, $a(i)$, $b(j)$, $a(i;j)$, $b(i;j)$, $c(i;j)$ are the MA, AR, non-linear MA, NAR, non-linear cross terms respectively and p and q denotes the order of the MA and AR process respectively. A simplified NARMA model is shown in Figure 1 where the system is represented by a linear process followed by a nonlinear function. This system can be compared to a Rayleigh channel, where $w(n)$ is the signal propagating through a faded and Additive White Gaussian Noise (AWGN) channel, system is the Rayleigh and AWGN channel and $h(n)$ is the faded signal at the receiver end.

B. Adaptive Filtering

Adaptive filtering is an effective method and has its implementations in a wide range of field, for example in system identification [20], [21] or equalizing a faded signal [22], [23] in wireless transmissions. The Finite Impulse Response (FIR) structure of an adaptive filter is shown in Figure 2. The following considerations are made,

For time instant n, let

$x(n)$ be the input to the adaptive filter;
$d(n)$ be the desired signal;
$\hat{d}$(n) be the adaptive filter output;
$e(n)$ be difference between the desired signal and the filter output;
p be the order of the filter; and
μ be the step size of the filter.

The filter coefficients are updated by Least Mean Square (LMS) algorithm [24].

$$e(n) = d(n) - \widehat{d}(\mathrm{n}) \ldots (2)$$

$$e(n) = d(n) - \sum_{t=0}^{t=p-1} \omega(t) - x(n-t) \ldots\ldots (3)$$

The filter coefficients are updated as,

$$\omega(n+1) = \omega(n) + \mu x(n) e(n)^{*} \ldots\ldots\ldots\ldots\ldots\ldots (4)$$

C. Prony's Method

Prony's method is based on the autocorrelation information carried by a signal and transmitted through a faded channel with some correlation characteristics. A Rayleigh faded channel can therefore be equalized by Prony's method. If $(p;q)$ is the order of the filter and a_p , b_q represent the poles and zeroes of the system, eq. 5 (Normal Eq.s) calculates the poles and eq. 7, solves the eq. for the zeroes. $r_x(k;l)$ represents the autocorrelation matrix of the input signal $x(n)$ for shifts k and l respectively as shown in eq. 6. The minimum value to model the error, $\varepsilon_{p,q}$ is shown in eq. 8. The error is modeled by minimizing the minimum mean square error [24].

Normal Eq.s:

$$\sum_{l=1}^{p} a_p(l) r_x(k,l) = -r_x(k,0); k = 1,2 \ldots p \ \ldots (5)$$

$$r_x(k,l) = \sum_{n=q+1}^{\infty} x(n-l)\,x^{*}(n-k); k,l \geq 0 \ldots (6)$$

Numerator:

$$b_q(n) = x(n) + \sum_{k=1}^{p} a_p\,(k) x(n-k);\ n = 1,2 \ldots q \ldots (7)$$

Minimum Error:

$$\epsilon_{p,q} = r_x(0,0) + \sum_{k=1}^{p} a_p(k)\, r_x(0,k) \ldots (8)$$

III. PROPOSED WORK

This section discuss in details about the proposed NARMA equalizer with a double feedback structure. It is divided into three subsections. Subsection 3.1 describes the proposed model, Subsection 3.2 discuss NARMA as a system identifier and Subsection 3.3 describes NARMA as an equalizer.

A. Proposed Model

Here, we consider a Single Input Single Output (SISO) multipath block fading system. The system model is shown in Figure 3. A few symbols are used as a training sequence at the initial stage. The transmitted symbols are Binary Phase Shift Keying (BPSK) modulated which are propagated through the fading and AWGN

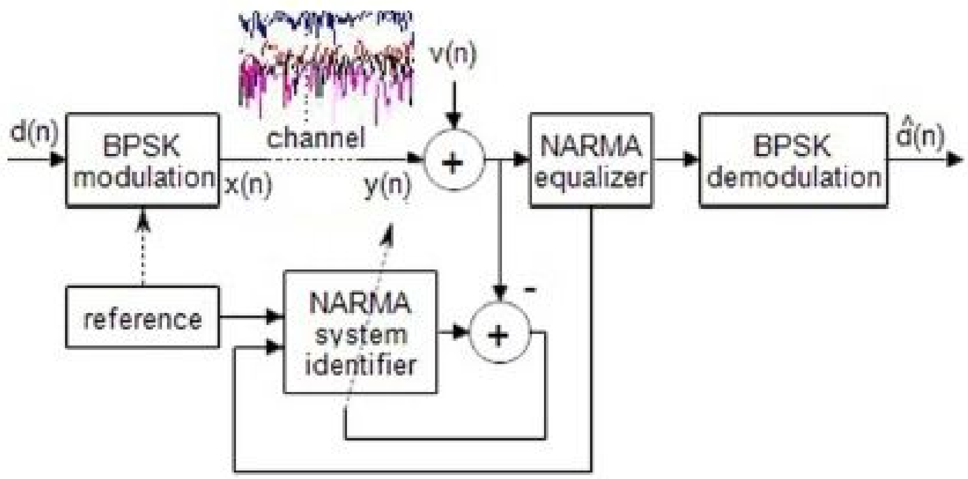

Figure 3: Proposed Model

channel. At the receiver end, the NARMA system identifier identifies the channel from some training symbols after some number of iterations determined by LMS algorithm. The NARMA equalizer uses the final coefficients obtained from the NARMA system identifier to further minimize the Mean Square Error (MSE) between the original and predicted symbols aided by LMS algorithm with a second feedback to the NARMA system identifier. The NARMA equalizer equalizes the symbols which are demodulated and decoded to get the transmitted bits.

From Figure 3 we make the following considerations,

For time instant n, let

$d(n)$ be the encoded bits;

$x(n)$ be the modulated symbols transmitted;

$y(n)$ be the received symbols;

$\hat{d}(n)$ be the decoded symbols;

$h^t(n)$ be channel coefficient for channel tap t;

and

$v(n)$ be the AWGN noise.

Ignoring the nonlinear terms, let the received signal be $\hat{y}(n)$ which can be represented as,

$$\hat{y}(n) = \sum_{\mathrm{t=0}}^{\mathrm{T-1}} h^t(n)\mathrm{x}(\mathrm{n}-1) + v(n) \dots\dots\dots (9)$$

To take the nonlinearities of the channel into account we consider a nonlinear function similar to 'tanh' function, represented as $\frac{1-e^{-x}}{1+e^{-x}}$ thus the nonlinear channel output can be written as,

$$\mathrm{Y(n)} = \frac{1-e^{-\widehat{y(n)}}}{1+e^{-\hat{y}(n)}} \dots\dots\dots\dots\dots\dots\dots\dots\dots\dots\dots (10)$$

B. NARMA system identifier

Figure 4 shows the structure of the NARMA system identifier. Here, we make the following considerations for time instant n, let $(p; q)$ be the order of the ARMA system identifier; $b_1, b_2, \dots b_q$ and $a_1, a_2, \dots a_p$ are the poles and zeros of the system respectively; $ref(n)$ represent the reference symbols for a particular frame; $arma(n)$ be the ARMA output; $narma(n)$ be the NARMA output; $e(n)$ be the error between the NARMA output and the channel output $y(n)$; μ_1, μ_2 be the step sizes for updating the LMS algorithm for poles and zeros respectively.

The ARMA coefficients are obtained by Prony's approximation as discussed in Subsection C of Section II. From eq. 10 the transfer function of the system can be represented as,

$$H(z) = \frac{arma\ (z)}{ref\ (z)} = \frac{\sum_{m=0}^{m=p} a_m z^{-m}}{1+\sum_{l=1}^{l=q} b_l z^{-m}} \dots\dots\dots\dots\dots\dots \dots(11)$$

Here, $H(z)$ is the z-transform of $h(n)$ and is related as,

$$H(z) = \sum_{n=0}^{n=\infty} h_n z^{-n} \dots\dots\dots\dots (12)$$

From eq. 11 we can write,

$$arma(z) = H(z)ref(z) \dots (13)$$

Eq. 11 can be expressed as a matrix multiplication and can be written in matrix form as,

$$\begin{pmatrix} a_0 \\ a_1 \\ a_2 \\ . \\ . \\ . \\ a_p \\ 0 \\ . \\ . \\ . \\ 0 \end{pmatrix} = \begin{pmatrix} h_0 & 0 & 0 & \dots & 0 \\ h_1 & h_0 & 0 & & \\ h_2 & h_1 & h_0 & & \\ . & & & & . \\ . & & & & . \\ . & & & & . \\ h_p & & & & . \\ . & & & & . \\ . & & & & . \\ h_k & \dots & & & h_{k-q} \end{pmatrix} \begin{pmatrix} 1 \\ b_1 \\ b_2 \\ . \\ . \\ . \\ b_q \end{pmatrix}$$

… (14)

The filter coefficients $a_1, a_2, \dots a_p$ and $b_1, b_2, \dots b_q$ are obtained by solving eq. 14.

Now, the ARMA output as seen from Figure 4 is given as,

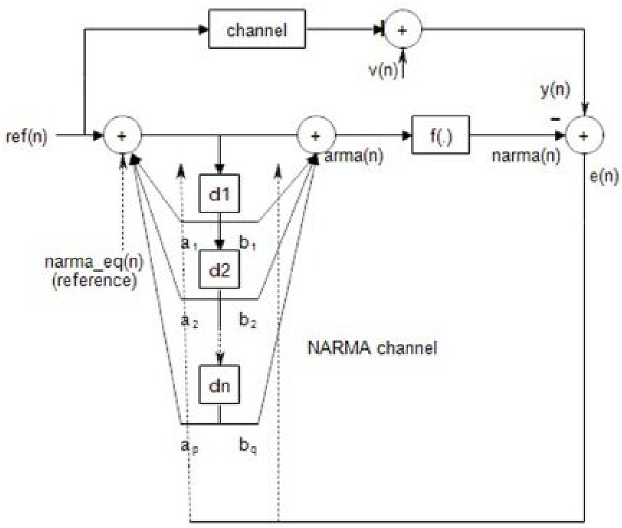

Figure 4: NARMA System Identifier

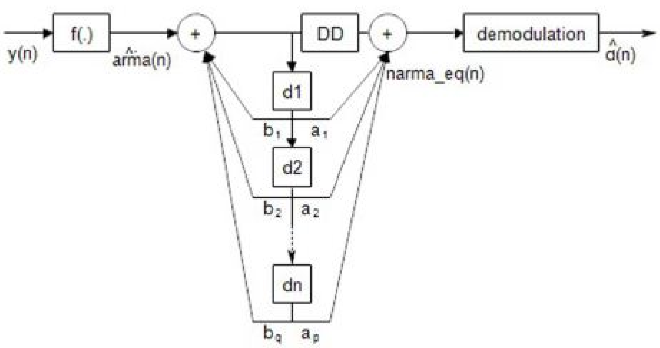

Figure 5: NARMA Equalizer

$$arma(n) = -\sum_{l=1}^{l=q} b_l \times arma(n-l) + \sum_{m=1}^{m=q} a_m \times ref(n-m) \ldots\ldots\ldots\ldots (15)$$

The NARMA output is given as,

$$narma(n) =) = \frac{1-e^{-arma(n)}}{1+e^{-arma(n)}} \ldots \ldots\ldots\ldots\ldots (16)$$

The error, $e1(n)$ required to update the coefficients is given as,

$$e1(n) = y(n) - narma(n) \qquad \ldots\ldots\ldots (17)$$

The poles and zeroes are updated by LMS algorithm as shown in eq.s 18 and 19 respectively,

$$b_l(n+1) = b_l(n) + \mu1(n) \times e1(n) \times narma(n) \ldots (18)$$

$$a_m(n+1) = a_l(m) + \mu2(n) \times e1(n) \times ref(n) \ldots (19)$$

As $e1$ (n) converges to some minimum value, the NARMA system tracks the channel effectively. Here μ_1 and μ_2 are the step functions.

C. NARMA-Double Feedback (NARMADF) Equalizer

The coefficients obtained from the NARMA system identifier is used by the NARMA equalizer to equalize the received signal. The structure of the equalizer is shown in Figure 5.

For time instant *n*, let *narma_ eq*(*n*) be the equalized bits. The received symbols are passed through a reverse non-linear process, whose output is given as,

$$\widehat{arma}(n) = log\frac{1-y(n)}{1+y(n)} \ldots\ldots\ldots\ldots .. (20)$$

The equalized symbols are denoted by,

$$narma\_eq(n) = \sum_{m=0}^{m=p} a_m \times narma\_eq(n-m) - \sum_{l=1}^{l=q} b_l \times \widehat{arma(n)}(n-l) \ldots \ldots\ldots\ldots (21)$$

During the training period these symbols are fed back to the NARMA system identifier as shown in Figure 5. Taking *narma eq*(*n*) as the new reference symbol to the NARMA system identifier block, from eq. 15, we obtain the NARMA system identifier output as *nar^ma*(*n*).

The NARMA feedback output is compared with the actual channel output, according to eq. 17 and if this value falls below a certain threshold set by the minimum value of absolute MSE obtained from the NARMA system identifier, the filter coefficients will update according to eq.s 18 and 19. As the error reduces, the final NARMA system identifier coefficients are used to determine the equalized symbols from the NARMA equalizer. This method thus provides a detailed training of the NARMA system and reduces of the MSE to some extent as compared to a NARMA-Single Feedback (NARMA-SF) equalizer. The steps for equalization using the proposed method is summarized in Algorithm 1.

IV. EXPERIMENTAL DETAILS AND RESULTS

This section describes in detail the results obtained by simulating the system under different channel conditions. In this section, the performance of the proposed system is compared with the results obtained from NARMA-SF in [5], ARMA and conventional methods like DFE and LS. The performance of the proposed method is measured in terms of Bit Error Rate (BER) plot. The entropy values of the decoded signal for both NARMA-DF and NARMA-SF are compared and shown in tabular form. The results validate the improvement in performance of the proposed system.

Here, we consider two different fading cases: pedestrian and vehicular for ITU specified channel conditions under 3GPP considerations. These channel parameters are illustrated in Table 1.The pedestrian case is assumed to have lesser number of channel taps than the vehicular case.

The BER-Signal to Noise Ratio (SNR) comparison plot for Case 1 is shown in Figure 6 and for Case 2, it is shown in Figure 7. A total of 105 symbols are generated and a training sequence of 25%

is taken for all the methods. Case 1 is a pedestrian case of 10*kmph* and 4 channel taps and thus fading in this case is lower than the vehicular case of 120*kmph* and 6 channel taps as in Case 2.

NARMA-DF gives a factor of 12, 80, 100, and 120 times improvement over NARMA-SF, ARMA, DFE and LS respectively for Case 1. The significantly high improvement in BER of NARMA-DF can be distinguished in Figure 7 for Case 2. A factor of 10, 200, 600 and 800 improvement occurs for NARMA DF over NARMA-SF, ARMA, DFE and LS respectively in this case. These values are averaged from the random BER for an SNR of 10*db.*

The filter parameters: order and length for all the methods under consideration; for both the pedestrian and vehicular cases are presented in Table 2.

Algorithm 1 Equalization by NARMA-DF

1: for each block of transmitted symbols do
2: for transmitted symbol = first reference symbol to last reference symbol do
3: Compute NARMA system identifier output from eq. 15 and 16.
4: Compute MSE from the error obtained from eq. 17.
5: Update NARMA system identifier coefficients from eq. 18 and 19.
6: end for
7: Find threshold from step 4.
8: end for
9: for each block of transmitted symbols do
10: for transmitted symbol = first data symbol to last data symbol do
11: Use final coefficients from step 5 and feed it to NARMA Equalizer along with received symbols.
12: Compute NARMA equalizer output from eq.s 20 and 21.
13: This predicted reference equalized output obtained from step 12 is fed into the NARMA System Identifier as the new reference symbol and steps 3 and 4 are repeated.
14: if error ϵ threshold then
15: Repeat step 5
16: end if
17: end for
18: Repeat step 11
19: for transmitted symbol = first reference symbol to last reference symbol do
20: Repeat step 12
21: Demodulate and decode the symbol.
22: end for
23: end for

Table1: Simulation Parameters

SL.NO	Parameters	Case 1	Case 2
1	Speed	10kmph	120kmph
2	Maximum Doppler shift	19.9Hz	238.89Hz
3	Number of Channel taps	4	6
4	Chip rate	3.84 x 10^6Hz	3.84 x 10^6Hz
5	Carrier frequency	2.15 x 10^9Hz	2.15 x 10^9Hz
6	Wavelength	0.139539m	0.139539m

Table 2: Filter parameters

Method	Case1		Case 2	
	Order (p,q)	Stepsize (μ1,μ2)	Order (p,q)	Stepsize (μ1,μ2)
DF NARMA	(4,2)	(0.05,0.001)	(6,2)	(0.05,0.001)
SF NARMA	(4,2)	(0.05,0.001)	(6,2)	(0.05,0.001)
ARMA	(4,2)	0.05	(6,2)	0.05
DFE	(4,2)	0.3	(6,2)	0.3

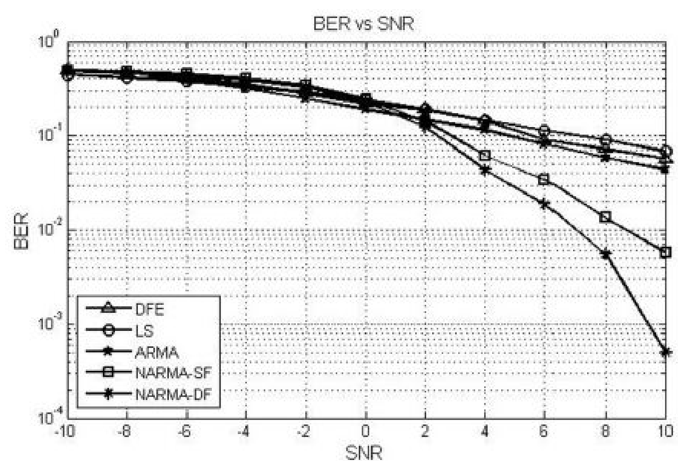

Figure 6: BER-SNR plot for Case 1

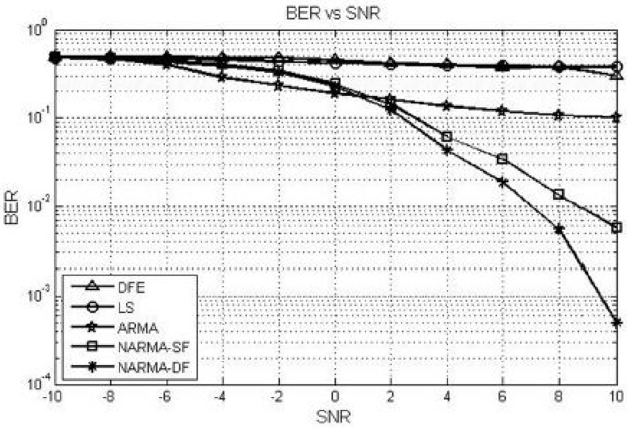

Figure 7: BER-SNR plot for Case 2

Table 3: BER values under different conditions for SNR= 20dB

Method		DF NARMA		SF NARMA	
Speed (kmph)	Taps	10% Pilot	25% pilot	10% pilot	25% pilot
30	4	0.0001	0.0009	0.0005	0.0028
	6	0.0037	0.0053	0.0051	0.0083
60	4	0.0049	0.0045	0.0170	0.0130
	6	0.0281	0.0051	0.0583	0.0094
90	4	0.0029	0.0001	0.0062	0.0027
	6	0.0247	0.0002	0.0682	0.0005
120	4	0.0024	0.0007	0.0047	0.0013
	6	0.0193	0.0071	0.0697	0.0147

Table 3 compares the BER for different number of channel taps and different training sequence lengths at an SNR of 20*db*. The results are shown for a channel length of 500 for different Doppler frequencies measured in terms of velocity of the receiver unit. These values are obtained from a random trial. From the Table 3, it can be inferred that with the increase in channel length, BER decreases for a particular Doppler ferquency, and minimizing the training overhead, compromises the BER to some extent. For 30*kmph* case, NARMA-DF gives a factor of 3:11 and 1:56 times improvement over NARMA-SF for 4 and 6 channel taps respectively for 25% training sequence and 5 and 1:4 times improvement over SF-NARMA for 4 and 6 channel taps respectively for 10% training sequence. For 60*kmph*, 90*kmph* and 120*kmph*, a BER improvement by a factor of 2, 2:4 and 3:6 times respectively is observed for DF-NARMA over SF-NARMA for 6 channel taps and 10% training sequence overhead condition.

Table 4 compares the random BER for LS, DFE and ARMA under different channel conditions. Here, LS gives the worst performance.

Table 4: Comparative BER values under different conditions at 20dB SNR

Method		LS	DFE	ARMA
Speed (kmph)	Taps	25%	25%	25%
30	4	0.0801	0.0759	0.0387
	6	0.4304	0.1629	0.1930
60	4	0.0628	0.0356	0.0176
	6	0.4049	0.0479	0.2486
90	4	0.0598	0.0347	0.0350
	6	0.4203	0.1468	0.0004
120	4	0.0441	0.0188	0.0285
	6	0.4407	0.1541	0.0029

Table 5: Entropy values at 20dB SNR under varying conditions

Method		DF NARMA		SF NARMA	
Speed (kmph)	Taps	10% pilot	25% pilot	10% pilot	25% pilot
30	4	1	1	0.99	0.99
	6	1	1	0.91	0.98
60	4	1	1	0.99	0.99
	6	1	1	0.91	0.98
90	4	1	1	0.99	0.99
	6	1	1	0.91	0.97
120	4	1	1	0.99	0.99
	6	1	1	0.90	0.99

Table 5 compares the information transmitted in terms of normalized entropy for the same conditions as in Table 3. The entropy increases to some extent for the proposed system. Also as the channel length increases for a particular Doppler frequency, fading rises, which decreases the amount of information transmitted and hence reduces the entropy. As shown in the table, NARMA-DF gives an average normalized increment of 0:1 and 0:9 units for a 25% and 10% training sequence respectively. The system overall gives a performance improvement by a factor of 10, 200, 600 and 800 for Case 2 when compared to SF-NARMA, ARMA, DFE and LS based equalization methods at an SNR of 10*db*.

Figure 8 compares the absolute MSE plot for a NARMA-SF and the resulting error after a feedback. From the figure, it can be concluded that the absolute error can be reduced by means of a feedback, and thus the probability of decoding the symbols correctly increases even for a short training sequence. Figure 8 is shown for a 10% training overhead for Case 2.

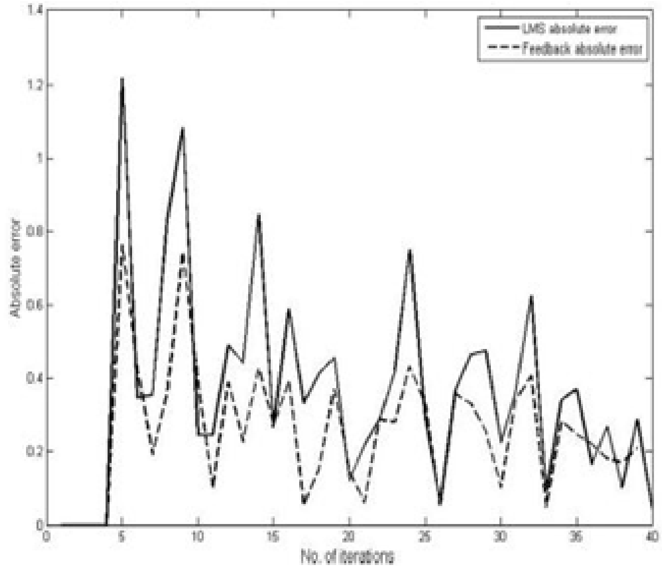

Figure 8: BER-SNR plot for Case 1

Table 6: Average accuracy in % demonstrated by NARMA (conventional), NARMA-SF and NARMA-DF (proposed)

SL no	SNR in dB	NARMA (conventional)	NARMA-SF	NARMA-DF
1	-10	84	88	89
2	-8	86	89	90
3	-6	86	90	91
4	-4	87	90	92
5	-2	87	90	92
6	0	88	91	93
7	2	88	92	94
8	4	89	92	95
9	6	90	93	95
10	8	90	93	95

Table 6 summarizes the accuracy performance in % demonstrated by NARMA (conventional), NARMA-SF and NARMA-DF (proposed). Between -10 and 10 dB, the accuracy in estimating the channel coefficient is between 89 to 95% which is better than both the NARMA (conventional) and NARMA-SF. The results are derived for twenty set of reading for vehicular speeds between 60 and 90 kmph. With double feedback, the proposed NARMA-DF proves to be an efficient tracker of the channel variations.

The results verify the performance improvement of the proposed system when compared to NARMA-SF, ARMA and conventional methods like DFE and LS. This is due to the detailed learning of the proposed method of double feedback structure. The proposed method could learn the finer nonlinear details of the channel during its training period and equalize the signal more efficiently for a short training sequence length. The proposed method is feasible to give better performance in severe environmental conditions with low training overheads. However, the major problem is the more computational time it takes to find the equalizer components due to the additional feedback. A balance is to be maintained between performance and delay observed.

V. CONCLUSION

Here, we discussed an approach to model the non-linear attributes of consider a mobile communication adopting a block faded SISO system. A NARMA based processing is used to mitigate the ill-effects due to fading which varies the quality of the received signal. A double feedback equalizer makes the training effective and improves the performance of the system which is ascertained by a number of BER v/s SNR curves with 3GPP considerations and ITU specifications. Performance variation is noted for different training overheads and a tradeoff between spectral efficiency and performance observed. The proposed equalizer structure proofs to be effective in severe fading conditions and works well with a low percentage of training symbols. In an extended and modified form the proposed design shall prove to be suitable for upcoming adaptive receiver designs as part of high data rate communication transceivers.

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