# Neuro Fuzzy System Based Adaptive Equaliser in Mobile Cellular Channels

SSNL Venkateswara Rao, Gottapu Sasibhushana rao

***Abstract: Mobile Cellular Channels are subjected to multi-path reflections by virtue of the environment they are operating. Owing to this, the transmitted symbols are subjected to different types of fading. This is more evident in urban scenario. The time-dispersive nature of the channel causes Inter symbol Interference (ISI). The Frequency re-use concept is adopted in mobile communications to raise the system capacity. This technique causes co channel interference (CCI). Spectral seepage in the system owing to filters, cause Adjacent channel interference. Hence the mobile channels are viewed as time varying. This ultimately results in more bit errors in the communication system. To keep the BER within the specified limits in time varying channels, receiver employs adaptive equalizer. These are based on different algorithms.. Most of these algorithms are proved to be much complex and not implementable in dealing with non linear and time varying channels. Of late, Neuro Fuzzy systems (NFS) are used to model adaptive equaliser. Fuzzy systems work on if then rules with membership functions (MF) to minimise output error. It is seen that the consolidation of fuzzy logic and neural network can deal efficiently with uncertainties associated with nonlinear time varying channels. In this paper, we present NFS based adaptive equaliser. Channel and Equaliser outputs are observed for the given input binary sequence. BER versus SINR curves are plotted and analysed.***

***Key words: Adaptive equaliser, ISI, the wireless channel, BER, the membership function***

## I. INTRODUCTION

The aim of any reliable communication system is to transfer data efficiently between Transmitter and Receiver. The reliability of Digital Communication (DC) systems over analog communication systems for transmitting data through the channel relies upon its inherent ability to resist the noise [1]. As a result, data can be transmitted at higher bit rates in DC systems than in Analog communication systems. In wireless communications, DC systems have a prime part. The channel between the transmitter and receiver also plays a pivotal role in reliable communication.

There is clear distinction between linear channel and mobile channel in communication systems. Linear channel with noise is time invariant [2].

However, mobile channels (MC) are time varying and non linear by virtue of its operating environment [3,4]. In MC systems, the signal transmitted from Base station is reflected from surrounding hilly terrain, buildings, clutter and other obstructions before they are intercepted at the receiver. As a result, multiple paths with varying propagation delays lead to the receiver. Speed of the mobile also gives rise to Doppler shift of the operating signal frequency [5]. As the channel characteristics are changing with time, these channels are known as time varying. Appropriate channel model is required to assess the technology for MC systems. Different channels are in vogue to model the MC channel. Some of the channel models are Tapped delay line (TDL), Free space, Rician fading, Raleigh fading, Hata okumura, FCC and ITU R model [6,7].TDL displays general structure of time varying channel. It is dealt with in Section 2. Interferences such as ACI, CCI and ISI tend to alter the transmitted data. This distorted data leads to increase bit error rate (BER) [8,9]. ISI arises because of time disseminative trait of the channel. Each pulse at the receiver is meddled with the adjoining pulses, there by distorting main pulse. In Mobile cellular channels, CCI and ACI are also contributing to the overall Interference [2,3]. These channels are further subjected to different types of fading. Fading causes random fluctuation in amplitude of the transmitted symbols. All these factors contribute to the nonlinear time varying nature of the mobile channels. Due to this phenomenon, significant bit errors occur in the transmitted symbols. For efficient and reliable communication, BER should be within specified limits. To reduce these bit errors, appropriate Equalizer is incorporated in the receiver.

Receivers use Equalisers to counter the ill effects of channel. Ideally, it offers the inverse response of the channel. Designing equalizers for Linear time invariant channels is relatively easier [8]. Designing of equalizers for MC channels is not a simple task. Adaptive filtering has a huge role in signal processing applications [ 9]. Earlier, linear adaptive equalizers were frequently used for its low cost implementation and simplicity. These equalizers are more suitable when channel properties are known before hand [9]. In time varying channels, the performance of linear equalizer is poor. Non linear signal processing techniques such as neural networks are employed in Bayesian equalizers [10]. Fuzzy filters are explored to understand their efficacy as non linear filters in equalization [8,9].

Equalisers based on fuzzy logic and neural network are yielding promising results

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while dealing with mobile channels . This paper is arranged in the ensuing manner [5]. Linear and Mobile channels with CCI are discussed in Section 2. General structure of Adaptive Equaliser with TDL is discussed in section 3. Generalised NFS structure is explored in Section 4. Section 5 gives results of Simulation. The Section 6 presents the conclusion.

## II. DIFFERENT COMMUNICATION CHANNELS

In this section, linear time invariant, linear time variant and Mobile cellular channels are discussed with equivalent channel models and equations.

### A. Linear Filter Channel (Time invariant)

The channels which are imitating the low pass filter response may be categorized as linear channels. They offer specified bandwidth. They do not interfere with other channels [3]. Telephone channels fall under this category [3]. The above channel is modeled as noise infusive linear channels , as shown in Fig. 1.

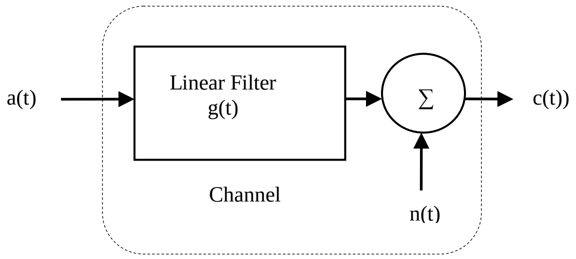


Fig. 1. The Linear Filter channel with additive noise

If a(t) is the input of the channel then the yield of the channel is

$$c(t) = a(t) * g(t) + n(t)$$
$$= \int_{-\infty}^{+\infty} g(\tau)a(t-\tau)d\tau + n(t) \quad (1)$$

* denotes convolution and g(t) is impulse response of the linear filter

### B. Linear Time variant Channel

The linear time-variant channel with additive noise is illustrated in Fig. 2. The transmitted signal in certain channels like ionosphere and acoustic channels is subjected to multi path propagation [10]. Those channels become time variant. Such channels are characterized by time-variant linear channels with impulse response g(t, τ). Parameter τ represents delay time variable.

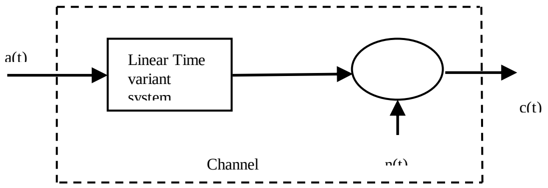


Fig 2. Linear Time variant channel with noise

The channel yield for a(t) as an input signal is

$$c(t) = a(t) * g(t,\tau) + n(t)$$
$$= \int_{-\infty}^{+\infty} g(t,\tau)a(t-\tau)d\tau + n(t) \quad (2)$$

A representation for multipath propagation in the mobile channels and ionosphere is obtained as a special case of Eqn (2) in which impulse response has the form

$$g(t,\tau) = \sum_{m=1}^{l} \alpha_m(t)\,\delta(\tau-\tau_m) \quad (3)$$

where the $\alpha_m(t)$ indicates the attenuation factors(time varying) for the l propagation paths. If eqn(2) is substituted into eqn(3), the signal at the receiver has the form

$$c(t) = \sum_{m=1}^{l} \alpha_m(t)\,a(t-\tau_m) + n(t) \quad (4)$$

Equation(4) comprises of l components, where each element is delayed by $\{\tau_m\}$ and decreased by $\{\alpha_m(t)\}$.

### C. Mobile Communication system model with CCI

Figure.3 displays the mobile communication system model with CCI[5]. The main channel along with n co-channels is depicted in Fig.3. Channel ouput is mixed with noise. The final channel output v(t) is applied to the Equaliser which is assumed to be at receiver end[4]. It is presumed that the receiver has the knowledge of the training signal. Various letter notations are as follows.

- n(t) is noise (white Gaussian),
- $g_j$ and $g_i$ are $j^{th}$ co channel and $i^{th}$ channel impulse responses
- d is the estimation delay [4]
- a(t) is symbol sequence at the transmitter
- v(t) is channel output
- $\hat{v}$(t-d) is an estimate of a(t)

The following equation expresses the received signal sequence

$$v(t) = a(t) + a_{cci}(t) + n(t) \quad (5)$$

Signals a(t) and $a_{cci}$(t) are related as follows.

$$a_{cci}(t) = \sum_{j=1}^{k}\sum_{i=1}^{n-1} g_j(i)\,a_i(t-i) \quad (6)$$

$$a(t) = \sum_{0}^{n} g_i\,a(t-i) \quad (7)$$

The co-channel, noise and desired signal samples are approximated to be not related. The signal to noise ratio (SNR) and the signal to interference noise ratio (SINR) and are defined as

$$SNR = \frac{\rho^2{}_s}{\rho^2{}_e} \quad (8)$$

$$SINR = \frac{\rho^2{}_s}{\rho^2{}_{cci}} \quad (9)$$

Here $\rho^2{}_s$ , $\rho^2{}_{cci}$ and $\rho^2{}_e$ are the signal power, the co-channel signal power and the noise power respectively [6].


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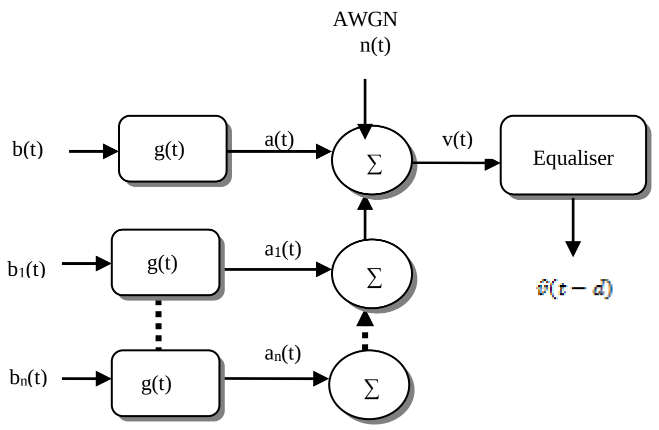


Fig. 3. Mobile cellular system with CCI

## III. ADAPTIVE EQUALISER

In this section, Adaptive equalization technique is discussed. When the knowledge of the channel is not available, the adaptive equalizer is more suitable [6].

An adaptive equalizer is a network that follows the characteristics of the time-varying channel. It is tuned consistently to match with time varying characteristics of channel. The transfer function of the above equaliser is changed as per optimizing algorithm. These algorithms are being widely utilised in signal processing applications [5]. The diagram of Adaptive equalizer is depicted in Fig.4.

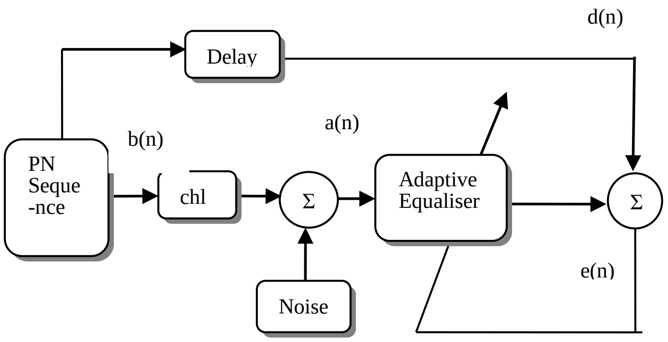


Fig. 4 Structure of Adaptive Equalizer with the channel.

Here the input signal is provided by Pseudo Noise Generator. Noise is added to the channel output as shown [11]. The adaptive equalizer tends to reduce the interferences caused by the channel.

It offers the inverse response of the channel so as to reduce the distortions . The function of adaptive algorithm is to minimise the cost function such as mean square error. We may adjust the variable parameters of the filter to achieve this. In our discussion, we consider tapped-delay line structure. This structure forms a finite impulse response structure with good stability. Tapped-delay line filter structure is shown in Fig.5.

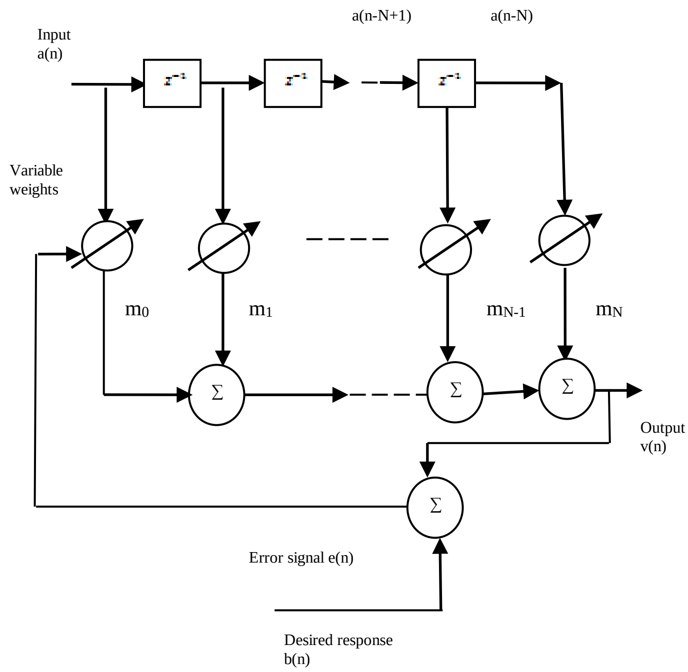


Fig.5 Block diagram of Tapped delay Line Equaliser

It is also known as transversal filter or Adaptive equalizer. It comprises of N number of delay elements, N+1 taps and N+1 variable weight. The location of the weight decides its value. The adaptive algorithm adjusts the equalizer coefficients based on the error and minimizes the cost function. The coefficients are updated either block by block basis or sample by sample basis. The difference between output v(n) and desired response b(n) gives the error. Here $z^{-1}$ indicates the delay component and m(i) is a multiplicative gain within the system . A single input a (n) is fed to the AE at any instant of time. The input a (n) of the AE is affected by noise and current state of the channel. Hence, a (n) is a random process.

## IV. GENERAL NEURO FUZZY STRUCTURE

The techniques based on Neural networks and Fuzzy logic, are adopted in the design of Intelligence systems. Neural networks are based on the structure of the human brain. Fuzzy logic exposes the way the brain deals with vague information. The common feature of both of them is to enhance the efficiency of intelligent systems in noisy and uncertain channels. Fuzzy systems and neural networks provide structured frame work and learning ability. NNs can perform complex operation between input and output. Multi layer perceptron, radial basis function and Recurrent neural network are some of the NN structures which are successful in equalization problems.

Figure 6 gives the general structure of NFS based Adaptive equalizer. Basically, it is forward feed


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configuration [12]. When the mathematical analysis is difficult, then the Fuzzy model is adopted to have the transmitted symbol estimation at the receiver [5]. It contains five layers as shown [11]. The rule base involves the following Takagi and Sugeno's if-then rules [11]. In the following description, R indicates rule and L indicates layer.

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R1 ➞ If a is C1 and b is D1, then $v1 = p1a + q1b + e1$

R2 ➞ If a is C2 and b is D2, then $v2 = p2a + q2b + e2$

## A. Layer description

- L1 – Each node n is a square node in this layer with a function

$$M_n^1 = \mu C_n(a) \tag{10}$$

where a – input to the node n,

$C_n$ – linguistic label

$M_n^1$—membership function of $C_n$ which is Bell shape function

$$\mu C_n(a) = \frac{1}{1+\left[\left(\frac{a-s_n}{l_n}\right)^2\right]^{m_n}} \tag{11}$$

Where ($l_n$, $m_n$, $s_n$) are parameters that decide the Bell shape function

- L2 – Each node gives the product of input signals

$$u_n = \mu C_n(a).\mu D_n(b), n = 1,2 \tag{12}$$

Equation (12) represents firing strength (FS)

- L3—This gives the ratio of $n^{th}$ rule FS to sum of all FSs

$$\overline{u_n} = \frac{u_n}{u_1+u_2}, n = 1,2 \tag{13}$$

- L4—Each node is expressed as a function given by

$$M_n^4 = \overline{u_n} v_n = \overline{u_n}(p_n a + q_n b + e_n) \tag{14}$$

Where $\overline{u_n}$ is L3 output and ($p_n, q_n, e_n$) is the parameter set

- L5—All input signals of this node are added

$$M_n^5 = \sum_n \overline{w_n}\, v_n = \frac{\sum_n w_n v_n}{\sum_n w_n} \tag{15}$$

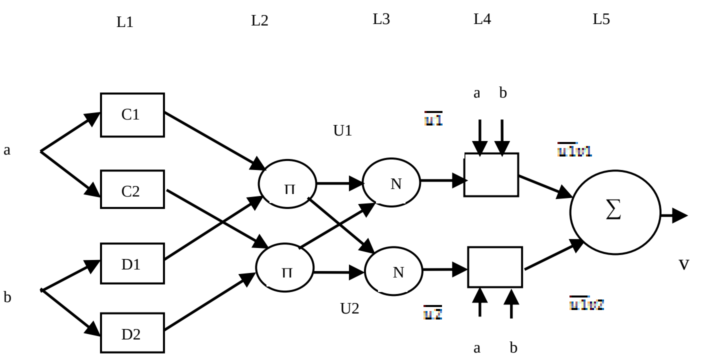


Fig.6 General Structure of Neuro Fuzzy Systems based Adaptive Equaliser

## V. SIMULATION RESULTS

Simulation results for above NFS Adaptive equaliser are displayed in Fig.7. Random binary sequence with 2048 bits of amplitude $\pm 1$ is generated. These RBS bits are given as Input to the channel. Mobile cellular channel with six co channel interference signals and white noise is taken. Threshold voltage for detector output is taken as $\pm$0.6v. The waveforms of channel input, channel output, Detector outputs of NFS 13 and NFS 17 adaptive equalisers are plotted in Fig.7. In NFS 13 adaptive equaliser, one input and three MFs are taken. In NFS 17 equaliser, same input and seven MFs are taken.

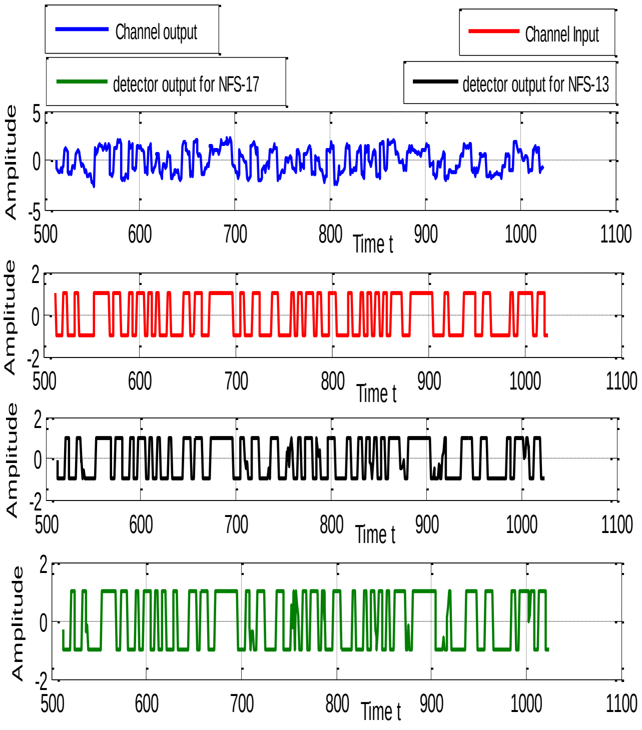


Fig.7 Results of channel input, channel output, Detector outputs of NFS equaliser

Values of log of SINR and log of BER are tabulated for NFS 13 and NFS17 Adaptive Equalisers in Table.1 and Table.2 respectively.

Table.1 Log of SINR and BER values for NFS 13 Adaptive equaliser

| | | | | | | | |
|---|---|---|---|---|---|---|---|
| SINR | -1.45 | 1.923 | 5.296 | 6.982 | 8.669 | 10.35 | 12.04 |
| BER | -0.273 | -0506 | -0.908 | -1.184 | -1.523 | -2.138 | -3.041 |

Table.2 Log of SINR and BER values for NFS 17 Adaptive equalizer

| | | | | | | | |
|---|---|---|---|---|---|---|---|
| SINR | -4.822 | -1.45 | 1.923 | 3.609 | 5.296 | 6.982 | 8.669 |
| BER | -0.083 | -0.278 | -0.557 | -0.767 | -1.05 | -1.564 | -2.342 |

Log of SINR versus log of BER values are plotted in Fig.8. RBS of 1100 bits are used in this simulation.


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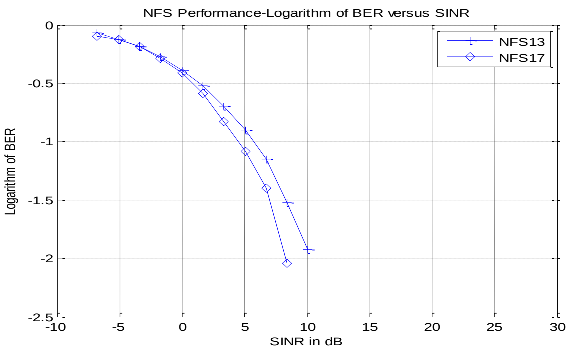


Fig.8 SINR versus BER curves for NFS 13 and NFS 17 Adaptive Equaliser

## VI. CONCLUSION

In this paper, NFS based Adaptive Equaliser is presented. Different channel models are presented before discussing equalisation. The design of Adaptive Equalisers for time varying channels such as Mobile cellular channels is relatively complex. The Channel output is distorted due to the effects of time varying characteristics of the channel, CCI and AWGN in Mobile channels. Modelling based on conventional mathematical methods and algorithms is not appropriate for MC channels. Soft computing methods can be employed to implement equalisers for time varying channels. The proposed configuration provides the benefits of neural network and Fuzzy logic. It is observed that the channel output is improved significantly after applying NFS equaliser. In the proposed method, three and seven MFs are utilised. It is observed that the BER values are decreasing almost exponentially with increasing values of SINR. Further, it can be inferred from the above plot that the BER curve is improving if the number of MFs is increased from three to seven for the same input. The structure of the NFS equaliser and type of learning algorithm can be explored to further improvise the above equaliser.

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