International Journal of Smart Home
Vol.4, No.4, October, 2010

Performance analysis of equalization techniques for MIMO systems in wireless communication

V.Jagan Naveen
Assoc Professor,
Dept of E.C.E,
GMR Institute of Technology,
Rajam.India,Mobile:+919849722092,
jagannaveen801@gmail.com

K.Murali Krishna
Professor
Dept of ECE, Sanketika vidyaparishat college of Engg,
Visakhapatnam, India,
Mobile:+91-9490041142,
mkkasi@yahoo.com

K.RajaRajeswari
Professor
Dept of ECE,
A.U College of Engineering,
Andhra University,
Visakhapatnam, India
Mobile:+91-9849238069,
krrau@hotmail.com

Abstract

In mobile communications systems, data transmission at high bit rates is essential for many services such as video, high quality audio and mobile integrated service digital network. When the data is transmitted at high bit rates, over mobile radio channels, the channel impulse response can extend over many symbol periods, which leads to Inter-symbol interference (ISI). Orthogonal Frequency Division Multiplexing (OFDM) is one of the promising technology to mitigate the ISI. In an OFDM signal the bandwidth is divided into many narrow sub-channels which are transmitted in parallel. Each sub-channel is typically chosen narrow enough to eliminate the effect of delay spread. By combining OFDM with CDMA dispersive fading limitations of the cellular mobile radio environment can be overcome and the effects of co-channel interference can be reduced. In this paper, the performances of equalization techniques by considering 2 transmit 2 receive antenna case (resulting in a 2×2 MIMO channel). Assume that the channel is a flat fading Rayleigh multipath channel and the modulation is BPSK.

The ultimate goal is to provide universal personal and multimedia communication without regard to mobility or location with high data rates. To achieve such an objective we need a strong equalization techniques to compensate ISI.

- *Zero Forcing (ZF) equalization*
- *Minimum Mean Square Error (MMSE) equalization*
- *Zero Forcing equalization with Successive Interference Cancellation (ZF-SIC)*
- *ZF_SIC with optimal ordering and*
- *MIMO with MMSE SIC and optimal ordering*

Index Terms— fading channels, inter symbol interference, Raleigh fading, adaptive equalizations.

1. Introduction

During the past few years, there has been an explosion in wireless technology. This growth has opened a new dimension to future wireless communications whose ultimate goal is to provide universal personal and multimedia communication without regard to mobility or location with high data rates. To achieve such an objective, the next generation personal communication networks will need to be support a wide range of services which will include high quality voice, data, facsimile, still pictures and streaming video. These future services are likely to include applications which require high transmission rates of several Mega bits per seconds (Mbps).

The data rate and spectrum efficiency of wireless mobile communications have been significantly improved over the last decade or so. Recently, the advanced systems such as 3GPP LTE and terrestrial digital TV broadcasting have been sophisticatedly developed using OFDM and CDMA technology. In general, most mobile communication systems transmit bits of information in the radio space to the receiver**.** The radio channels in mobile radio systems are usually multipath fading channels, which cause inter-symbol interference (ISI) in the received signal. To remove ISI from the signal, there is a need of strong equalizer which requires knowledge on the channel impulse response (CIR).[1]

Equalization techniques which can combat and/or exploit the frequency selectivity of the wireless channel are of enormous importance in the design of high data rate wireless systems. Although such techniques have been studied for over 40 years, recent developments in signal processing, coding and wireless communications suggest the need for paradigm shifts in this area.

On one hand, the demonstrated efficiency of soft-input soft-output signal processing algorithms and iterative (turbo) techniques have fuelled interest in the design and development of nearly optimal joint equalization and decoding techniques. On the other hand, the popularity of MIMO communication channels, rapidly time varying channels due to high mobility, multi-user channels, multi-carrier based systems and the availability of partial or no channel state information at the transmitter and/or receiver bring new problems which require novel equalization techniques. [2]

Hence, there is a need for the development of novel practical, low complexity equalization techniques and for understanding their potentials and limitations when used in wireless communication systems characterized by very high data rates, high mobility and the presence of multiple antennas.[10]

In radio channels, a variety of adaptive equalizers can be used to cancel interference while providing diversity. Since the mobile fading channel is random and time varying, equalizers must track the time varying characteristics of the mobile channel, and thus are called *adaptive equalizers*.

The general operating modes of an adaptive equalizer include *training* and *tracking*. First, a known, fixed-length training sequence is sent by the transmitter so that the receiver's equalizer may adapt to a proper setting for minimum bit error rate (BER) detection. The training sequence is typically a pseudorandom binary signal or a fixed, prescribed bit pattern. Immediately following this training sequence, the user data (which may or may not include coding bits) is sent and the adaptive equalizer at the receiver utilizes a recursive algorithm to evaluate the channel and estimate filter coefficients to compensate for the distortion created by multipath in the channel. The training sequence is designed to permit an equalizer at the receiver to aquire the proper filter coefficients in the worst possible channel conditions(e.g., fastest velocity, longest time delay spread, deepest fades, etc.) so that when the training sequence is finished, the filter coefficients are near the optimal values for reception of user data. As user data are received, the adaptive algorithm of the equalizer tracks the changing channel. As a consequence, the adaptive equalizer is continually changing its filter characteristics over time. When an equalizer has been properly trained, it is said to have converged.

The time span over which an equalizer converges is a function of the equalizer algorithm, the equalizer structure, and the time rate of change of the multipath radio channel.[3]

2. MIMO-OFDM Systems

A MIMO-OFDM system with four transmit and p(p≥4) receive antennas is shown in Fig.2.1. Though the figure shows MIMO-OFDM with four transmit antennas, the techniques developed in this paper can be directly applied to OFDM systems with any number of transmit antennas.[4]

At time , each of two data blocks,$b_i[n,k]$: k= 0,1….for i=1 and 2, is transformed into two different signals,$\{t_{2(i-1)+j}[n,k]$: k =0,1….., j=1,2$\}$for i= 1 and 2,respectively, through two space–time encoders. The OFDM signal for the th transmit antenna is modulated by $t_i[n,k]$ at the kth tone of the nth OFDM block.[9]

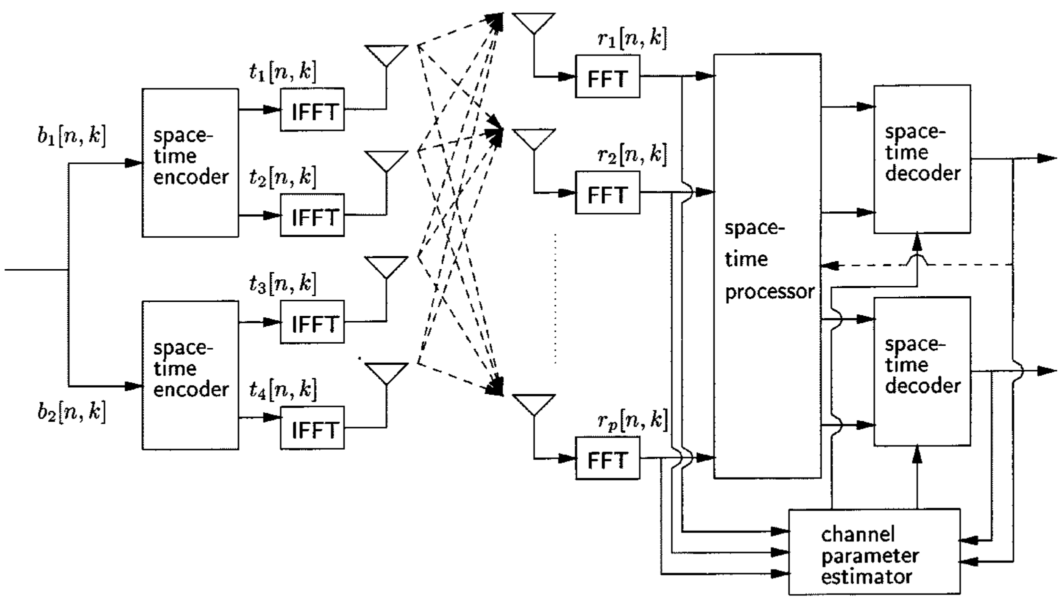

Figure 2.1 MIMO OFDM system

From the figure, the received signal at each receive antenna is the superposition of four distorted transmitted signals, which can be expressed as

$$\sum_{i=1}^{4} H_{ij}[n,k]\, t_i[n,k] + w_j[n,k] \ldots\ldots\ldots\ldots\ldots (1)$$

For j=1…….p . $w_j[n,k]$ in (1) denotes the additive complex Gaussian noise at the j th receive antenna, and is assumed to be zero-mean with variance σ_n^2 and uncorrelated for different n's,k 's, or j's. $H_{ij}[n,k]$ in (1) denotes the channel frequency response for the kth tone at n

time , corresponding to the ith transmit and the j th receive antenna. The statistical characteristics of wireless channels are briefly described in Section II-B.
The input–output relation for OFDM can be also expressed in vector form as

$$r[n,k]= H_1[n,k]t_1[n,k]+H_2[n,k]t_2[n,k]+w[n,k]$$

where,

$$r[n,k] \triangleq \begin{pmatrix} r1[n,k] \\ . \\ rp[n,k] \end{pmatrix} \qquad w[n,k] \triangleq \begin{pmatrix} w1[n,k] \\ . \\ wp[n,k] \end{pmatrix}$$

$$t_i[n,k] \triangleq \begin{pmatrix} t(2i-1)[n,k] \\ t(2i)[n,k] \end{pmatrix}$$

and

$$h_i[n,k] \triangleq \begin{pmatrix} H2i-1[n,k] & h2i[n,k] \\ \vdots & \vdots \\ h2i-1p[n,k] & h2ip[n,k] \end{pmatrix}$$

To achieve transmit diversity gain and detect the transmitted signal, a space–time processor must extract the required signals for space–time decoders. Note that both the space–time processor and space–time decoding require channel state information.

3. Adaptive Equalizer

An equalizer is usually implemented at the baseband or at IF in a receiver. Since the baseband complex envelope expression can be used to represent band pass waveforms, the channel response, demodulated signal and adaptive equalizer algorithms are usually simulated and implemented at the baseband.

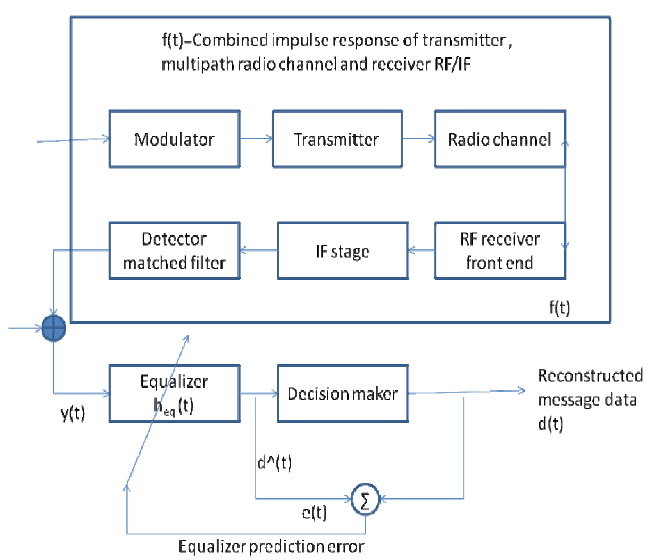

Figure 3.1 block diagram of a communication system with an adaptive equalizer in the receiver

Fig 3.1 shows a block diagram of a communication system with an adaptive equalizer in the receiver. If x(t) is the original information signal and f(t) is the combined complex baseband impulse response of the transmitter, channel and RF/IF sections of the receiver, signal received by equalizer can be expressed as

$$Y(t) = x(t) \otimes f^*(t) + n_b(t)$$

Where $f^*(t)$ denotes complex conjugate of f (t), $n_b(t)$ is the baseband noise at the input of the equalizer, and * denotes the convolution operation. If the impulse response of the equalizer is $h_{eq}(t)$, then the output of equalizer is

$$d(t) = x(t) \otimes f^*(t) \otimes h_{eq}(t) + n_b(t) \otimes h_{eq}(t)$$

$$= h(t) \otimes g(t) + n_b(t) \otimes h_{eq}(t)$$

Where g(t) is the combined response of the transmitter , channel ,RF/IF sections of the receiver ,and the equalizer at the receiver. The complex base band impulse response of the transversal filter equalizer is given by

$$h_{eq}(t) = \sum_n C_n \delta(t - nT)$$

where c_n are the complex filter coefficients of the equalizer. The desired output of the equalizer is x(t) ,the original source data. Assume that $n_b(t)$. Then, in order to force d (t) = x (t) in equation 3.2,g(t) must be equal to

$$g(t) = f^*(t) \otimes h_{eq}(t) = \delta(t)$$

The goal of equalization is to satisfy the given equation .

$$H_{eq}(f)\, F^*(-f) = 1$$

Where $H_{eq}(f)$ and F (f) are Fourier transforms of $h_{eq}(t)$ and f(t) respectively.
so that the combination of the transmitter , channel , and receiver appear to be an all pass channel. [7]

4.1 Zero forcing (ZF) equalizer

The math for extracting the two symbols which interfered with each other. In the first time slot, the received signal on the first receive antenna is,

$$y_1 = h_{1,1}x_1 + h_{1,2}x_2 + n_1 = [h_{1,1} \; h_{1,2}]\begin{bmatrix} x_1 \\ x_2 \end{bmatrix} + n_1.$$

The received signal on the second receive antenna is,

$$y_2 = h_{2,1}x_1 + h_{2,2}x_2 + n_2 = [h_{2,1} \; h_{2,2}]\begin{bmatrix} x_1 \\ x_2 \end{bmatrix} + n_2.$$

Where

y_1, y_2 are the received symbol on the first and second antenna respectively,

$h_{1,1}$ is the channel from 1^{st} transmit antenna to 1^{st} receive antenna,

$h_{1,2}$ is the channel from 2^{nd} transmit antenna to 1^{st} receive antenna,

$h_{2,1}$ is the channel from 1^{st} transmit antenna to 2^{nd} receive antenna,

$h_{2,2}$ is the channel from 2^{nd} transmit antenna to 2^{nd} receive antenna,

x_1, x_2 are the transmitted symbols and

n_1, n_2 is the noise on 1^{st}, 2^{nd} receive antennas.

We assume that the receiver knows $h_{1,1}$, $h_{1,2}$, $h_{2,1}$ and $h_{2,2}$. The receiver also knows y_1 and y_2. The unknown s are x_1 and x_2. Two equations and two unknowns

For convenience, the above equation can be represented in matrix notation as follows:

$$\begin{bmatrix} y_1 \\ y_2 \end{bmatrix} = \begin{bmatrix} h_{1,1} & h_{1,2} \\ h_{2,1} & h_{2,2} \end{bmatrix} \begin{bmatrix} x_1 \\ x_2 \end{bmatrix} + \begin{bmatrix} n_1 \\ n_2 \end{bmatrix}$$

Equivalently,

$$\boldsymbol{y} = \boldsymbol{H}\boldsymbol{x} + \boldsymbol{n}$$

To solve for $\boldsymbol{x}$, we know that we need to find a matrix $\boldsymbol{W}$ which satisfies $\boldsymbol{W}\boldsymbol{H} = \boldsymbol{I}$. The Zero Forcing (ZF) linear detector for meeting this constraint is given by,

$$\boldsymbol{W} = (\boldsymbol{H}^H \boldsymbol{H})^{-1} \boldsymbol{H}^H.$$

This matrix is also known as the pseudo inverse for a general m x n matrix.

The term,

$$H^H H = \begin{bmatrix} h_{1,1}^* & h_{1,2}^* \\ h_{2,1}^* & h_{2,2}^* \end{bmatrix} \begin{bmatrix} h_{1,1} & h_{1,2} \\ h_{2,1} & h_{2,2} \end{bmatrix}$$

Note that the off diagonal terms in the matrix $H^H H$ are not zero . Because the off diagonal terms are not zero, the zero forcing equalizer tries to null out the interfering terms when performing the equalization, i.e when solving for x_1 the interference from x_2 is tried to be nulled and vice versa. While doing so, there can be amplification of noise. Hence Zero Forcing equalizer is not the best possible equalizer to do the job. However, it is simple and reasonably easy to implement.[5]

For BPSK modulation in Rayleigh fading channel, the bit error rate is derived as,

$$P_b = \frac{1}{2}\left(1 - \sqrt{\frac{(E_b/N_0)}{(E_b/N_0)+1}}\right).$$

4.2 Minimum Mean Square Error (MMSE) equalizer

The math for extracting the two symbols which interfered with each other. In the first time slot, the received signal on the first receive antenna is,

$$y_1 = h_{1,1}x_1 + h_{1,2}x_2 + n_1 = [h_{1,1}\ \ h_{1,2}]\begin{bmatrix} x_1 \\ x_2 \end{bmatrix} + n_1.$$

The received signal on the second receive antenna is,

$$y_2 = h_{2,1}x_1 + h_{2,2}x_2 + n_2 = [h_{2,1}\ \ h_{2,2}]\begin{bmatrix} x_1 \\ x_2 \end{bmatrix} + n_2.$$

where

y_1, y_2 are the received symbol on the first and second antenna respectively,

$h_{1,1}$ is the channel from 1^{st} transmit antenna to 1^{st} receive antenna,

$h_{1,2}$ is the channel from 2^{nd} transmit antenna to 1^{st} receive antenna,

$h_{2,1}$ is the channel from 1^{st} transmit antenna to 2^{nd} receive antenna,

$h_{2,2}$ is the channel from 2^{nd} transmit antenna to 2^{nd} receive antenna,

x_1, x_2 are the transmitted symbols and

n_1, n_2 is the noise on 1^{st}, 2^{nd} receive antennas.

We assume that the receiver knows $h_{1,1}$, $h_{1,2}$, $h_{2,1}$ and $h_{2,2}$. The receiver also knows y_1 and y_2. For convenience, the above equation can be represented in matrix notation as follows:

$$\begin{bmatrix} y_1 \\ y_2 \end{bmatrix} = \begin{bmatrix} h_{1,1} & h_{1,2} \\ h_{2,1} & h_{2,2} \end{bmatrix} \begin{bmatrix} x_1 \\ x_2 \end{bmatrix} + \begin{bmatrix} n_1 \\ n_2 \end{bmatrix}$$

Equivalently,

$$\boldsymbol{y} = \boldsymbol{H}\boldsymbol{x} + \boldsymbol{n}$$

The Minimum Mean Square Error (MMSE) approach tries to find a coefficient $\boldsymbol{W}$ which minimizes the criterion,

$$E\left\{[\boldsymbol{W}\boldsymbol{y}-\boldsymbol{x}][\boldsymbol{W}\boldsymbol{y}-\boldsymbol{x}]^H\right\}.$$

Solving,

$$\boldsymbol{W} = \left[\boldsymbol{H}^H\boldsymbol{H} + N_0\boldsymbol{I}\right]^{-1}\boldsymbol{H}^H.$$

When comparing to the equation in Zero Forcing equalizer , apart from the $N_0\boldsymbol{I}$ term both the equations are comparable. Infact, when the noise term is zero, the MMSE equalizer reduces to Zero Forcing equalizer .[6]

4.3 Zero Forcing with Successive Interference Cancellation (ZF-SIC)

Using the Zero Forcing (ZF) equalization approach described above, the receiver can obtain an estimate of the two transmitted symbols x_1, x_2, i.e.

$$\begin{bmatrix} \hat{x}_1 \\ \hat{x}_2 \end{bmatrix} = (\boldsymbol{H}^H\boldsymbol{H})^{-1}\boldsymbol{H}^H \begin{bmatrix} y_1 \\ y_2 \end{bmatrix}.$$

Take one of the estimated symbols (for example $\hat{x}_2$) and subtract its effect from the received vector y_1 and y_2, i.e.

$$\begin{bmatrix} r_1 \\ r_2 \end{bmatrix} = \begin{bmatrix} y_1 - h_{1,2}\,\hat{x}_2 \\ y_2 - h_{2,2}\,\hat{x}_2 \end{bmatrix} = \begin{bmatrix} h_{1,1}\,x_1 + n_1 \\ h_{2,1}\,x_1 + n_2 \end{bmatrix}.$$

Expressing in matrix notation,

$$\begin{bmatrix} r_1 \\ r_2 \end{bmatrix} = \begin{bmatrix} h_{1,1} \\ h_{2,1} \end{bmatrix} x_1 + \begin{bmatrix} n_1 \\ n_2 \end{bmatrix}$$

$$\boldsymbol{r} = \boldsymbol{h} x_1 + \boldsymbol{n}$$

The above equation is same as equation obtained for receive diversity case. Optimal way of combining the information from multiple copies of the received symbols in receive diversity case is to apply Maximal Ratio Combining (MRC).

The equalized symbol is,

$$\hat{x}_1 = \frac{\boldsymbol{h}^H \boldsymbol{r}}{\boldsymbol{h}^H \boldsymbol{h}}.$$

This forms the simple explanation for Zero Forcing Equalizer with Successive Interference Cancellation (ZF-SIC) approach.

4.4 Successive Interference Cancellation with optimal ordering

In classical Successive Interference Cancellation, the receiver arbitrarily takes one of the estimated symbols, and subtract its effect from the received symbol $y_1$and y_2. However, we can have more intelligence in choosing whether we should subtract the effect of $\hat{x}_1$first or $\hat{x}_2$first. To make that decision, let us find out the transmit symbol (after multiplication with the channel) which came at higher power at the receiver. The received power at the both the antennas corresponding to the transmitted symbol x_1 is,

$$P_{x_1} = |h_{1,1}|^2 + |h_{2,1}|^2.$$

The received power at the both the antennas corresponding to the transmitted symbol x_2 is,

$$P_{x_2} = |h_{1,2}|^2 + |h_{2,2}|^2.$$

If $P_{x_1} > P_{x_2}$then the receiver decides to remove the effect of $\hat{x}_1$from the received vector $y_1$and y_2 and then re-estimate $\hat{x}_2$.

$$\begin{bmatrix} r_1 \\ r_2 \end{bmatrix} = \begin{bmatrix} y_1 - h_{1,1}\ \hat{x}_1 \\ y_2 - h_{1,2}\ \hat{x}_1 \end{bmatrix} = \begin{bmatrix} h_{1,2}\ x_2 + n_1 \\ h_{2,2}\ x_2 + n_2 \end{bmatrix}$$

Expressing in matrix notation,

$$\begin{bmatrix} r_1 \\ r_2 \end{bmatrix} = \begin{bmatrix} h_{1,2} \\ h_{2,2} \end{bmatrix} x_2 + \begin{bmatrix} n_1 \\ n_2 \end{bmatrix}$$

$$\boldsymbol{r} = \boldsymbol{h} x_2 + \boldsymbol{n}$$

Optimal way of combining the information from multiple copies of the received symbols in receive diversity case is to apply Maximal Ratio Combining [4] (MRC). The equalized symbol is,

$$\hat{x}_2 = \frac{\boldsymbol{h}^H \boldsymbol{r}}{\boldsymbol{h}^H \boldsymbol{h}}.$$

Else if $P x_1 \leq P x_2$ the receiver decides to subtract effect of $\hat{x}_2$ from the received vector y_1 and y_2, and then re-estimate $\hat{x}_1$

$$\begin{bmatrix} r_1 \\ r_2 \end{bmatrix} = \begin{bmatrix} y_1 - h_{1,2}\ \hat{x}_2 \\ y_2 - h_{2,2}\ \hat{x}_2 \end{bmatrix} = \begin{bmatrix} h_{1,1}\ x_1 + n_1 \\ h_{2,1}\ x_1 + n_2 \end{bmatrix}$$

Expressing in matrix notation,

$$\begin{bmatrix} r_1 \\ r_2 \end{bmatrix} = \begin{bmatrix} h_{1,1} \\ h_{2,1} \end{bmatrix} x_1 + \begin{bmatrix} n_1 \\ n_2 \end{bmatrix}$$

$$\boldsymbol{r} = \boldsymbol{h} x_1 + \boldsymbol{n}$$

Optimal way of combining the information from multiple copies of the received symbols in receive diversity case is to apply Maximal Ratio Combining [4] (MRC). The equalized symbol is,

$$\hat{x}_1 = \frac{\boldsymbol{h}^H \boldsymbol{r}}{\boldsymbol{h}^H \boldsymbol{h}}.$$

Doing successive interference cancellation with optimal ordering ensures that the reliability of the symbol which is decoded first is guaranteed to have a lower error probability than the other symbol. This results in lowering the chances of incorrect decisions resulting in erroneous interference cancellation. Hence gives lower error rate than simple successive interference cancellation.[10]

4.5 MMSE SIC and optimal ordering

We extend the concept of successive interference cancellation to the MMSE equalization and simulate the performance. We will assume that the channel is a flat fading Rayleigh multipath channel and the modulation is BPSK.[11][12]

Brief description of 2×2 MIMO transmission, assumptions on channel model and the noise are detailed in on Minimum Mean Square Error (MMSE) equalization.

4.5a MMSE equalizer for 2×2 MIMO channel

Let us now try to understand the math for extracting the two symbols which interfered with each other. In the first time slot, the received signal on the first receive antenna is,

$$y_1 = h_{1,1}x_1 + h_{1,2}x_2 + n_1 = [h_{1,1} \ h_{1,2}]\begin{bmatrix} x_1 \\ x_2 \end{bmatrix} + n_1.$$

The received signal on the second receive antenna is,

$$y_2 = h_{2,1}x_1 + h_{2,2}x_2 + n_2 = [h_{2,1} \ h_{2,2}]\begin{bmatrix} x_1 \\ x_2 \end{bmatrix} + n_2.$$

:

$$\begin{bmatrix} y_1 \\ y_2 \end{bmatrix} = \begin{bmatrix} h_{1,1} & h_{1,2} \\ h_{2,1} & h_{2,2} \end{bmatrix}\begin{bmatrix} x_1 \\ x_2 \end{bmatrix} + \begin{bmatrix} n_1 \\ n_2 \end{bmatrix}.$$

Equivalently,

$$\boldsymbol{y} = \boldsymbol{H}\boldsymbol{x} + \boldsymbol{n}$$

The Minimum Mean Square Error (MMSE) approach tries to find a coefficient $\boldsymbol{W}$ which minimizes the criterion,

$$E\left\{[\boldsymbol{W}\boldsymbol{y} - \boldsymbol{x}][\boldsymbol{W}\boldsymbol{y} - \boldsymbol{x}]^H\right\}.$$

Solving,

$$\boldsymbol{W} = \left[\boldsymbol{H}^H\boldsymbol{H} + N_0\boldsymbol{I}\right]^{-1}\boldsymbol{H}^H.$$

Using the Minimum Mean Square Error (MMSE) equalization, the receiver can obtain an estimate of the two transmitted symbols x_1, x_2, i.e.

$$\begin{bmatrix} \hat{x}_1 \\ \hat{x}_2 \end{bmatrix} = (\boldsymbol{H}^H\boldsymbol{H} + \boldsymbol{N}_0\boldsymbol{I})^{-1}\boldsymbol{H}^H\begin{bmatrix} y_1 \\ y_2 \end{bmatrix}.$$

4.5b Successive Interference Cancellation

In classical Successive Interference Cancellation, the receiver arbitrarily takes one of the estimated symbols (for example the symbol transmitted in the second spatial dimension, $\hat{x}2$), and subtract its effect from the received symbol $y1$and $y2$. Once the effect of $\hat{x}2$is removed, the new channel becomes a one transmit antenna, 2 receive antenna case and can be optimaly equalized by Maximal Ratio Combining (MRC).

However, we can have more intelligence in choosing whether we should subtract the effect of $\hat{x}1$first or $\hat{x}2$first. To make that decision, let us find out the transmit symbol (after multiplication with the channel) which came at higher power at the receiver. The received power at the both the antennas corresponding to the transmitted symbol $x1$ is,

$$Px_1 = |h_{1,1}|^2 + |h_{2,1}|^2.$$

The received power at the both the antennas corresponding to the transmitted symbol $x2$ is,

$$Px_2 = |h_{1,2}|^2 + |h_{2,2}|^2.$$

If $Px_1 > Px_2$then the receiver decides to remove the effect of $\hat{x}1$from the received vector $y1$and y_2 $\hat{x}2$. Else if $Px_1 \leq Px_2$the receiver decides to subtract effect of $\hat{x}2$from the received vector $y1$and $y2$, and then re-estimate $\hat{x}1$.

Once the effect of either $\hat{x}1$or $\hat{x}2$is removed, the new channel becomes a one transmit antenna, 2 receive antenna case and the symbol on the other spatial dimension can be optimally equalized by Maximal Ratio Combining (MRC).For detailed equations on the construction of the new 2 x 1 channel using successive interference cancellation, please refer to the post on ZF-SIC with optimal ordering.[3]

5. Results

5.1 Zero forcing equalizer

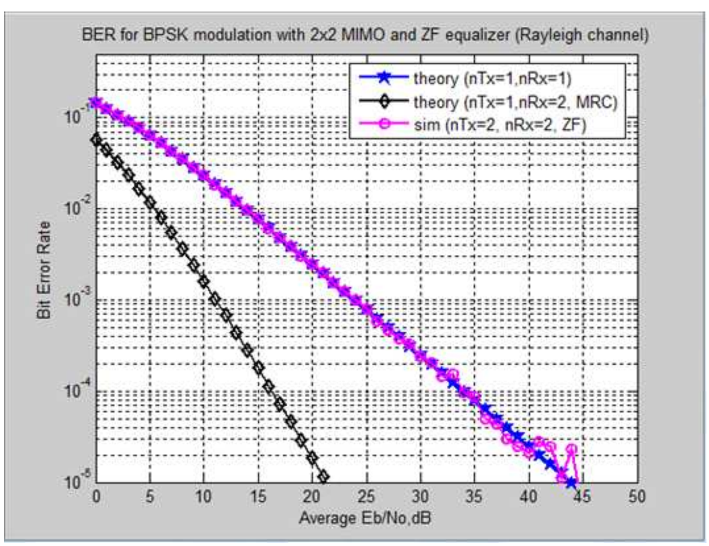

Figure5.1: BER plot for 2×2 MIMO channel with ZF equalizer
(BPSK modulation in Rayleigh channel)

The simulated results with a 2×2 MIMO system using BPSK modulation in Rayleigh channel is showing matching results as obtained in for a 1×1 system for BPSK modulation in Rayleigh channel. The zero forcing equalizer helps us to achieve the data rate gain, but NOT take advantage of diversity gain.[8]

5.2 Minimum Mean Square Error (MMSE) equalization

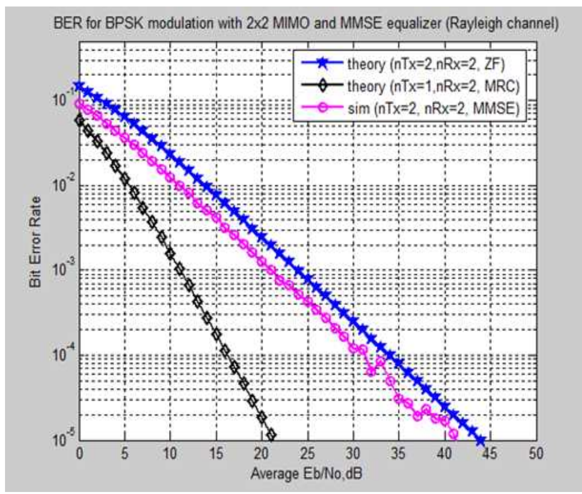

Figure 5.2: BER plot for 2×2 MIMO with MMSE equalization for BPSK in Rayleigh channel.As Compared to the Zero Forcing equalizer case, at 10-3 BER point, it can be seen that the Minimum Mean Square Error (MMSE**)** equalizer results in around 3dB of improvement.

5.3. Zero Forcing equalization with Successive Interference Cancellation (ZF-SIC)

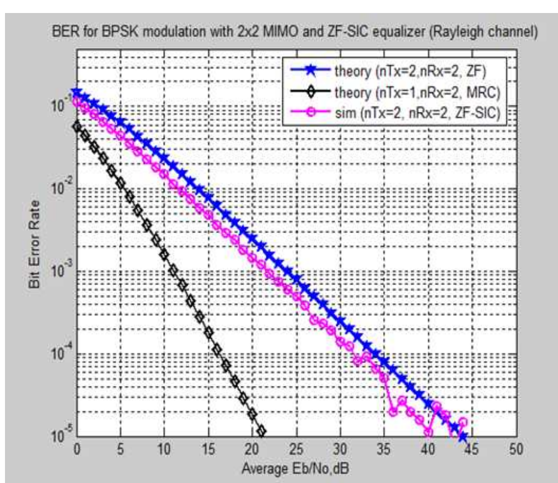

Figure 5.3: BER plot for BPSK in 2×2 MIMO channel with Zero Forcing Successive Interference Cancellation equalization

As Compared to Zero Forcing equalization alone case, addition of successive interference cancellation results in around 2.2dB of improvement for BER of 10^{-3}.The improvement is brought in because decoding of the information from the first spatial dimension (x1) has a lower error probability that the symbol transmitted from the second dimension.

5.4. ZF-SIC with optimal ordering

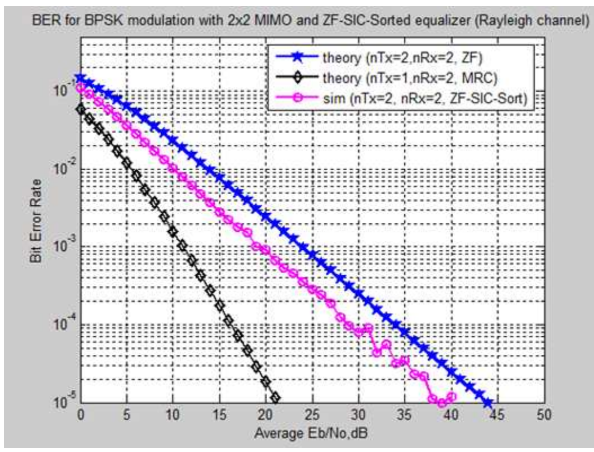

Figure 5.4: BER plot for BPSK in 2×2 MIMO equalized by ZF-SIC with optimal ordering

Compared to Zero Forcing equalization with successive interference cancellation case, addition of optimal ordering results in around 2.0dB of improvement for BER of 10^{-3}. Successive interference cancellation with optimal ordering ensures that the reliability of the symbol which is decoded first is guaranteed to have a lower error probability than the other symbol. This results in lowering the chances of incorrect decisions resulting in erroneous interference cancellation.

5.5. MMSE SIC and optimal ordering

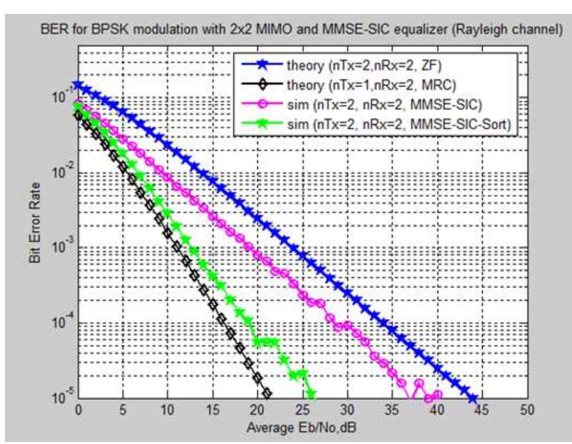

Figure 5.5: BER plot for 2×2 MIMO channel with MMSE-SIC equalization with and without optimal ordering

Compared to Minimum Mean Square Equalization with simple successive interference cancellation case, addition of optimal ordering results in around **5.0dB of** improvement **f**or BER of 10^{-3}.

6. Conclusion

Equalisation techniques are of enormous importance in the design of high data rate wireless systems. They can combat for inter symbol interference even in mobile fading Channel with high efficiency. In this paper performance of different equalization techniques has been analysed to find out suitable equaliser for 2x2 MIMO channel in Rayleigh multipath fading environment.

Zero Forcing equaliser performs well only in theoretical assumptions that are when noise is zero. Its performance degrades in mobile fading environment.

Minimum Mean Square Error (MMSE**)** equalizer uses LMS (Least Mean Square) as criterion to compensate ISI. The MMSE equalizer results in around 3dB of improvement when compared to zero forcing equalizer.

Zero forcing with Successive interference cancellation improves the performance of equalizer. This process improves the estimator performance on the next component compared to the previous one. Compared to Zero Forcing equalization alone case, addition of successive interference cancellation results in around 2.2dB of improvement for BER.

Zero forcing with Successive interference cancellation with optimal ordering ensures that the reliability of the symbol which is decoded first is guaranteed to have a lower error probability than the other symbol. Compared to Zero Forcing equalization with successive interference cancellation case, addition of optimal ordering results in around 2.0dB of improvement for BER.

Minimum Mean Square Equalization with simple successive interference cancellation case, addition of optimal ordering results in around 5.0dB of improvement for BER.

So, by observing the simulation results we conclude that by using MMSE with SIC optimal ordering, interference can be cancelled at optimum level even in a mobile fading channel.

References

[1] [DIG-COMM-BARRY-LEE-MESSERSCHMITT] Digital Communication: Third Edition, by John R. Barry, Edward A. Lee, David G. Messerschmitt

[2] [WIRELESS-TSE, VISWANATH] Fundamentals of Wireless Communication, David Tse, Pramod Viswanath

[3] R. Scholtz, "Multiple Access with Time-Hoping Impulse Modulaton," IEEE milit. Commun. Conf., vol . 2, pp. 447-450,1993.

[4] Wireless communications and networks : second edition, by Theodore S. Rappaport

[5] "ZERO-FORCING EQUALIZATION FOR TIME-VARYING SYSTEMS WITH MEMORY " by Cassio B. Ribeiro, Marcello L. R. de Campos, and Paulo S. R. Diniz.

[6] ZERO-FORCING FREQUENCY DOMAIN EQUALIZATION FOR DMT SYSTEMS WITH INSUFFICIENT GUARD INTERVAL by Tanja Karp , Martin J. Wolf , Steffen Trautmann , and Norbert J. Fliege

[7] "Adaptive Equalization" by SHAHID U. H. QURESHI, SENIOR MEMBER, IEEE

[8] Approximate Minimum BER Power Allocation for MIMO Spatial Multiplexing Systems Neng Wang and Steven D. Blostein, Senior Member, IEEE
[9] MIMO-OFDM modem for WLAN by, Authors: Lisa Meilhac, Alain Chiodini, Clement Boudesocque, Chrislin Lele, Anil Gercekci.
[10] G. Leus, S. Zhou, and G. B. Giannakis, "Orthogonal multiple access over time- and frequency-selective channels," IEEE Transactions on Information Theory, vol. 49, no. 8, pp. 1942–1950, 2003.
[11] P. A. Bello, "Characterization of randomly time-variant channels," IEEE Transactions on Communications, vol. 11, no. 4, pp. 360–393, 1963.
[12] Wireless communications and Networking by VIJAY GARG.

Authors

v.Jagan Naveen is currently working as a Associate Professor in ECE Department G M R Institute of Technology, Rajam, India. He is working towards his PhD at AU College of Engineering, Vishakhapatnam, India. He received his M.E from Andhra University Engineering college, vishakapatnam, India. His research interests are in the areas wireless communications and signal processing.

Murali Krishna Kasi was born in Kakinada, A P, India, on November 15, 1970. He received the (B.Tech) degree in electronics and communication engineering from N.U in 1996. The ME in 2001 and Ph.D., in 2008, from Andhra University. He is now professor and Department Head, Sanketika Vidya Parishadh College of Engineering. His area of interest is OFDM-MIMO, MC-CDMA, and OFDMA Systems.

K. Raja Rajeswari obtained her BE ME and PhD degrees from Andhra University, Visakhapatnam, India in 1976, 1978 and 1992 respectively. Presently she is working as a professor in the Department of Electronics and Communication Engineering, Andhra University. She has published over 100 papers in various National, International Journals and conferences. She is Author of the textbook *Signals and Systems* published by PHI. She is co-author of the textbook *Electronics Devices and Circuits* published by Pearson Education. Her research interests include Radar and Sonar Signal Processing, Wireless CDMA communication technologies etc. She has guided *ten* PhDs and presently she is guiding *twelve* students for Doctoral degree. She is current chairperson of IETE, Visakhapatnam Centre. She is recipient of prestigious IETE Prof SVC Aiya Memorial National Award for the year 2009, Best Researcher Award by Andhra University for the year 2004 and Dr. Sarvepalli Radhakrishnan Best Academician Award of the year by Andhra University for the year 2009. She is expert member for various national level academic and research committees and reviewer for various national/international journals.