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SCHOLARLY ARTICLE

Minimizing the Destructive Effects of Multipath Signal Propagation on Wireless Radio Signal

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ABSTRACT

Signals transmitted through wireless mobile channels suffer from fading, signal dispersion and distortion resulting in communication errors such as inter symbol interference. The atmospheric layers and obstructions have a significant effect on the behavior of signals as they propagate within the channel. This research was aimed at determining an efficient adaptive equalization method that can minimize the destructive effects of multipath signals in a wireless communication system. To achieve this objective, a Raleigh fading channel model was developed and an adaptive algorithm that mimics the desired filter by finding the filter coefficients that produce the least mean square of the error signal was also developed. The source recovery error was used to define a performance or cost function that corrects the effect of delay to recover the main source information. The decision-directed and dispersion minimization algorithms were developed to minimize the communication errors resulting from the multipath signal. effects. Both adaptive equalization methods had a great ripple reduction but the decision-directed equalizer has a very fast convergence speed, greater steady-state error and scored optimal values that are within the MSE value of 10^(-2) as compared to dispersion minimization equalizer

Keywords: *Multipath Signal Propagation; Adaptive Equalizer; Decision-Directed Algorithm; Dispersion Minimization Algorithm*

Introduction

In wireless electronic communication, the radio signals are distorted as they travel from the transmitting end to the receiving end. The adverse effect of multipath is observed when the signals arrive at the receiving end with different phases thus giving rise to low-strength signals. This is because these out-of-phase signals tend to subtract rather than add to the receiver. This phenomenon is called destructive interference and it causes serious distortion to listeners. This destructive interference is Rayleigh in nature and causes signal cancellation which likely weakens and/or attenuates signal power below the recognizable threshold. The interference induced by the multiple copies, also called multipath waves, has become the most remarkable cause of distortion known as fading and inter-symbol interface (ISI). Some authors have done research works that were aimed at finding a solution to this problem. These include the works of Sunil Kumar et al, on Rician fading channels, using an M-PAM modulation scheme in a Simulink environment (Sunil et al 2013). It is evident in their work that when the Rician K factor is increased, the signal-to-noise ratio increases and as a result the bit error rate decreases.

Narashimha Murthy and Satyanarayana compared Rayleigh and Rician fading environment under a frequency selective fading environment Narashimha and Satyanarayana, (2010), They were able to show that the Rician fading model exhibited a smaller number of deep fades with stronger line-of-sight (LOS) components, reducing noise and interferences thereby improving the signal amplitude and quality. Mardeni and Priya Siva researched the path loss suffered by the Wimax signals Mardeni and Priya (2010). They compared the COST231 Hata model with the Egli model and Stanford university interim model and discovered that the COST231 Hata model determined the path loss suffered by the WiMAX signals more accurately which can be used by telecommunication providers to improve their services.

Tommi Heikkila presented a paper on the basics of the rake receiver technique, its implementation and its design in cellular systems Tommi (2004). The rake receiver collects the time-shifted version of the original signal by providing a separate correlation receiver for each of the multipath signals. The correlator outputs are combined to achieve improved communications reliability and performance. An adaptive equalization was developed by Garima and Amadeep using tapped delay line filter to minimize mean square interference Garima and Amadeep (2011). The main limitation of the technique is that convergence of the tap weight is assured only for relatively low dispersion channels.

Debashree and Durgesh adjusted the tap weights with its adaptive training algorithm and the convergence was assured for all channel response pulses Debashree & Durgesh (2009), The degree of suboptimality of the tap weight setting reached by the training algorithm may or not be consequential depending on the application. Its limitation was the non-consideration of the effect of the equalizer on the receiver noise so that the problem of sub optimally does not arise. Hani investigated the use of adaptive wiener filter and least mean square (LMS) algorithm in the design of channel equalizers Hani (2004). Based on the magnitude and phase responses, the LMS algorithm outperforms the Wiener filter by 5dB which shows a very significant performance gap. Prachi compared Zero Forcing (ZF) with LMS and Recursive Least Squares (RLS) algorithms for a linear adaptive equalizer. This was done for the need of an equalizer with a fast convergence rate. LMS performed better than ZF. RLS exhibit better performance than all but it is complex and unstable. Hence it is avoided in practical implementation. This work is geared toward determining an efficient adaptive equalization method that can reduce the destructive effects of multipath signals in a wireless communication system.

Theory

Fading Channels

Wireless communication has proven to be vital in our everyday lives. However, the performance of the wireless communication systems is often limited or corrupted due to the nature of the path between the transmitter and the receiver being severely obstructed by buildings and trees. Hence, a transmitted signal may travel through a path depending on the characteristics of the radio channel. Signals propagating in the radio channel experience fading. A fading channel is a communication channel that has to face different fading phenomena while the signal is being carried from the transmitter to the receiver. Fading channels confront a phenomenon called multipath which occurs when all the radio propagation effects combine in a real-world environment. That is to say, when multiple signal propagation paths exist, the received signal level is the vector sum of all signal's incident from any direction or angle of arrival. The total composite signal is thus called the multipath signal

Multipath Fading

As stated earlier, fading is a phenomenon that occurs as a result of sudden and spontaneous variation in signal strength across any propagation media. Multipath propagation contributes to the occurrence of fading in wireless communication. The phenomenon results in radio signals reaching the receiving antenna through more than one path. The destructive effects of multipath propagation can be described in terms of fading and delay spread of the signals Mohamed et al (2014). The real-world multipath occurs when there is more than one path available for radio propagation. The phenomenon of reflection, diffraction and scattering all give rise to additional radio propagation paths beyond the direct line of sight path. Under this situation, some signals will aid the direct path while others will subtract from the direct signal path. The total composite phenomenon thus results in what is called multipath fading. Multipath fading can affect signal transmission through flat fading and selective fading. In flat fading, all the frequency components are equally affected and it causes the amplitude to fluctuate over time. Selective fading ensures that the selected frequency components of the signals are affected thereby having increased error and attenuation as compared to other frequency components of the same signal.

Interference

Radio wave interference is the phenomenon that occurs when two waves meet while traveling along the same medium. The interference of waves causes the medium to take on a shape that results from the net effect of the two individual waves upon the particles of the medium. Mohamed et al considered two pulses of the same amplitude traveling in different directions along the same medium (Naftaly 2012). Suppose that each pulse is upward 1 unit at its crest and has the shape of a sine wave. As the sine pulse moves towards each other, there will eventually be a moment in time when they are completely overlapped. At that moment, the resulting shape of the medium would be an upward displaced sine pulse with an amplitude of 2 units.

Constructive interference is a type of interference that occurs at any location along the medium where the two interfering waves have a displacement in the same direction. In this case, both waves have upward displacement; consequently, the medium has upward displacement that is greater than the displacement of the two interfering pulses, constructive interference is observed at any location where the two interfering waves are displaced upward. But it is also observed when both interfering waves are displaced downwards.

Destructive interference is a type of interference that occurs at any location along the medium where the two interfering waves have a displacement in the opposite direction. For instance, when a sine pulse with a maximum displacement of +1 unit meets a sine pulse with a maximum displacement of -1-unit, destructive interference occurs.

Equalizer

An equalizer is a filter that is usually adjustable and is meant to compensate for the unequal frequency response of some other signal processing circuit or system. A typical equalizer allows the user to adjust one or more parameters that determine the overall shape of the filter's transfer function. It is generally used to improve the fidelity of sound, emphasize certain instruments, remove undesired noises, or create completely new and different sounds. The purpose of this equalization was to undo the adverse effects of the channel and remove interference (Prachi 2014) The equalizer attempts to build a system that is a "delay inverse" of the digital model of the transmission channel, removing the inter-symbols interference while simultaneously rejecting the additive interference that is uncorrelated to the source. An adaptive equalizer is a filter that automatically adjusts to the tune of the changing properties of the communication channel. It can be implemented to prefer tap-weight adjustments periodically or continually. Periodic adjustments are accomplished by periodically transmitting a preamble or short training sequence of digital data known by the receiver (Akaneme and Onoh 2015). Continual adjustments are accomplished by replacing the known training sequence with the sequence of digital data symbols estimated from the equalizer output and treated as known data. When performed continually and automatically in this way the adaptive procedure is referred to as decision directed. If the probability of error exceeds one percent, the decision-directed equalizer might not converge. A common solution to these problems is to initialize the equalizer with an alternate process, such as a preamble to provide channel error performance, and then switch to decision directed mode

Development of a Fading Channel Model

The model assumes that N equal strength rays arrive at a moving receiver with uniformly distributed arrival angles α_n, such that ray n experiences a Doppler shift $\omega_d = \omega_m \, cos\alpha_n$ where $\omega_m = 2\pi f v/c$ is the maximum Doppler frequency shift, v is the vehicle speed, f is the carrier frequency and c is the speed of light. Using $\alpha_n = 2\pi f v/c - \pi/N$ and $\beta_n = \pi n/N_O$, there is quadrantal symmetry in the magnitude of the doppler shift. As a result, the fading waveform can be modeled with $N_O = N/4$ complex oscillators and this leads to the following equation;

$$R_k(t) = \sqrt{\frac{2}{N_O}} \sum_{n=1}^{N_O} A_k(n)(cos\beta_n + jsin\beta_n)\cos(\omega_m cos\alpha_n t + \theta_n) \quad (1)$$

Where k=1, 2,…N_O. N_O is the number of distinct oscillators, β_n is the phase angle of impinging waves, A_k is the k^{th} Walsh Hadamard code word, α_n is the angle of arrival of n^{th} wave and θ_n is the n^{th} independent random phases uniformly distributed between 0 and 2π. The input parameters are the mobile speed $v = 100kmph$, the carrier frequency $f_c = 1800MHz$ for a typical GSM system, sample frequency $f_s = 10kbps$, the number of channel coefficients $u = 3$ and the number of sub-channels $M = 1000$. They were simulated in MATLAB to obtain the Raleigh fading channel.

Development of Equalization processes for Optimization of the Tap Coefficients

Least Mean Square Algorithm (LMS) is an adaptive algorithm that mimics the desired filter by finding the filter coefficients that produce the least mean square of the error signal. It is based on finite impulse response adaptive filtering where the filter coefficient corresponds to the weight vector of impinging signals on each array. The least mean square filter was built around a tap delay line structure as illustrated in figure 1

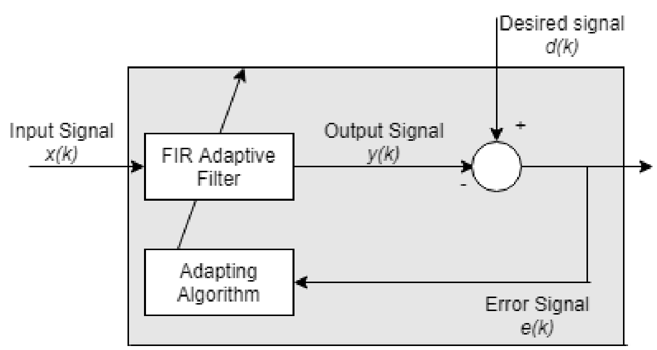

Figure 1: Adaptive Filter

The output equation is defined as
$y[k] = f[k].x[k]$ (2)
Where $f[k]$ is the filter coefficient or weight vector while $x[k]$ is the input of the signal vector. The error signal $e[k]$is the difference between the desired and actual signal and is defined as;

$$e[k] = d[k] - y[k] = d[k] - f[k].x[k] \quad (3)$$

Where $d[k]$is the desired signal
The equation for optimizing the tap coefficients is

$$f_{k+1} = f_k - \mu e_k \, {x^*}_k \text{ (} \qquad 4)$$

Where f_k denotes the estimate of the tap coefficient vector at the instant k, f_{k+1} is the update of the previous estimate, $\mu$is the step size, e_k is the error at the time and ${x^*}_k$ is the conjugate of the input signal to the equalizer at time k. the step-size parameter can be selected based on the rule of thumb to ensure convergence and good tracking by

$$0 < \mu < \frac{1}{5(2k+1)P_R} \qquad (5)$$

P_R is the received signal plus noise power

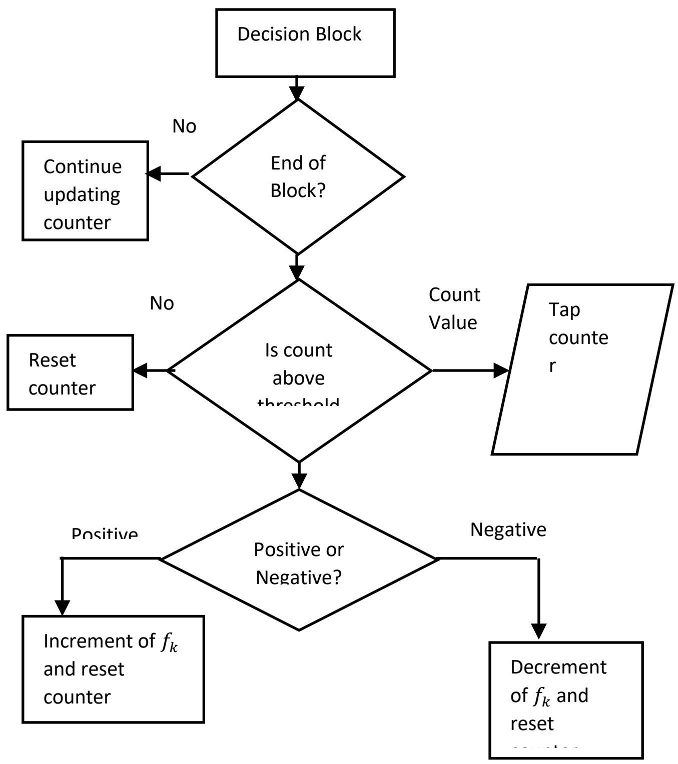

Figure 2: Coefficient Update Criteria for LMS

The flow chart in figure 2 illustrates the coefficient adjustment procedure that was followed to minimize the mean square error. The coefficient was updated on every iteration of the algorithm. If convergence was not reached, the system would continue to update the weight level. The threshold was checked on every iteration. If the count was above the threshold, a decision is taken and if otherwise, the count would reset to begin iteration. When the count was above the threshold and gives a positive value, the tap coefficient is increased but if it is a negative value, the tap coefficient decreases. It is the increment and decrement of the weight vector that compensates for the error to achieve equalization.

Determination of Source Recovery Error

The error due to the transmitted signal was estimated by sending the source signal in advance to the receiver. The error due to the delayed version of the source signal is given as;

$$e[k] = s[k - \partial] - y[k] \qquad (6)$$

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Where $s[k-\partial]$is the delayed version of the source signal. The source recovery error was used to define a performance or cost function that corrects the effect of delay to recover the original source information. The least-square cost function is given as; $J_{LS} = E^T E = (S - RF)^T (S - RF)$
$J_{LS} = S^T$[1-R $((R^T R)^{-1} R^T]S$ (7)

Development of Decision-Directed Algorithm for Adaptive Linear Equalizer

The decision-directed algorithm was developed for fast convergence speed and greater steady-state error. This will help to minimize the communication errors resulting from the multipath signal effects. The flow chart for the algorithm is shown in figure 3.

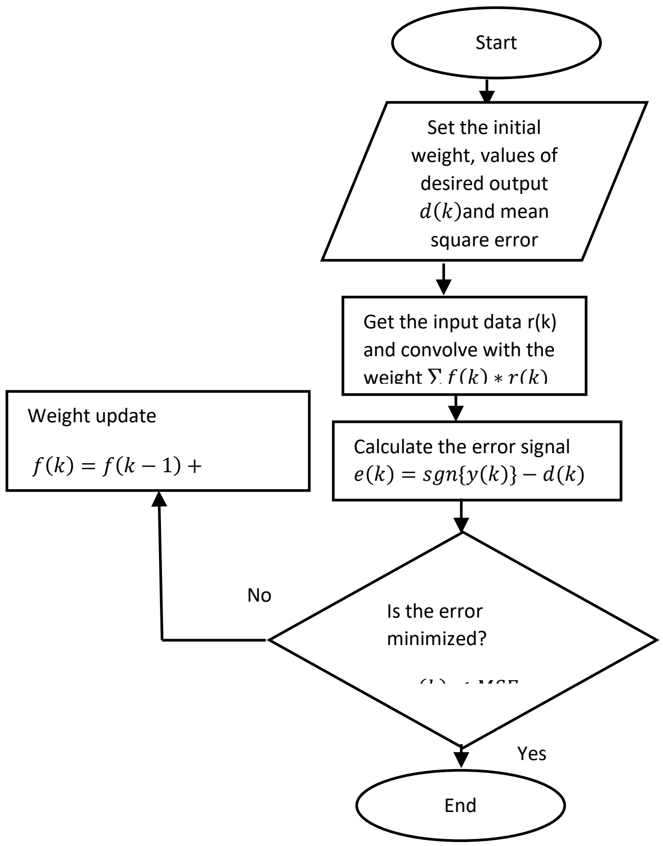

Figure 3: Decision-Directed LMS Algorithm

The weights are initialized, the error level and the desired output are being set. The weight multiplies the sampled number of time-delayed versions of an incoming input signal that gives the actual output as the summation of all input terms. The error signal results from the difference between the actual output and the expected or desired output signal. The decision block determines if the error is minimized or not. If the error is below the minimum, the iteration ends but if the error remains above the minimum, the iteration will continue to update the filter weight.

Development of Dispersion Minimization Equalization Algorithm

The dispersion minimization equalization algorithm was developed as shown in figure 4.

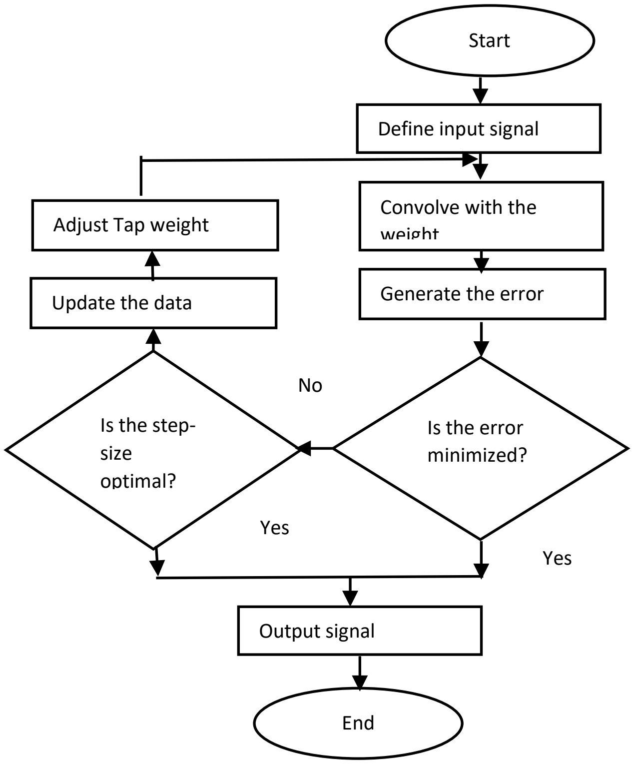

Figure 4: Dispersion Minimization Equalization Algorithm

The input signal is defined by initializing the weights and values of the desired output. The equalizer weight is convolved with the input sequence to produce the output sequence. The error is generated by comparing the input sequence with the reference signal. The error level will be checked if it is minimized at an optimal step size. If the error is below the minimum, the iteration ends but if the error remains above the minimum, the iteration will continue to update the coefficient. The weight vector is updated until convergence is obtained.

Result

The Raleigh fading channel developed is shown in figure 5. It showed a cluster of waveforms whose magnitude swing lies between 10dB and -35dB. The Doppler effect was high causing dense fluctuation and occurrence of signal outages at the intervals 100, 150, 170, 200, 220, 300, 450, 520, 600, 780 and 900 seconds. An increase in the speed of the mobile increases the level of fluctuation of the envelope. A higher frequency of the carrier signal also produces a higher inter symbol interference to the signal but an increase in the sampling frequency reduces the signal fluctuation.

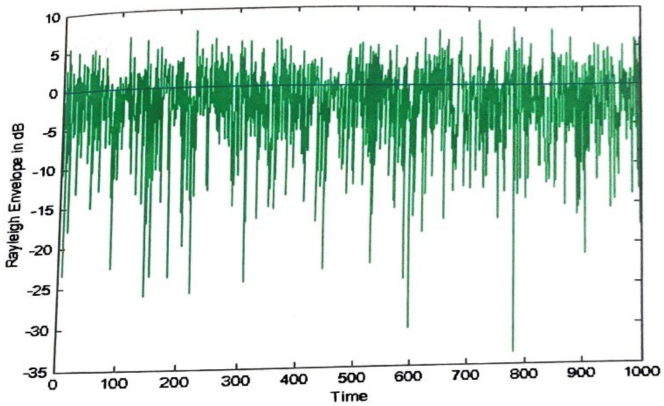

Figure 5: Raleigh Fading Channel

From the output of the channel, adaptive equalization techniques were used to minimize the communication errors such as the inter symbol interference that resulted from the destructive effect of the multipath signal created by the channel. By the creation of tap delay lines, this effect can be ameliorated. The performance function corrected the effect of the delay as shown in table 1.

Table 1: Minimum Performance Function

Delay (Delta)	*Error*	J_{LS}	*Equalization Coefficient f*
0	3400	786.362	0.3682, -0.0010, 0.0490, -0.0008
1	0	140.4086	0.6818, 0.3712, 0.618, 0.0821
2	0	30.2020	-0.2720, 0.6483, 0.3078, 0.1426
3	0	46.9299	0.1013, -0.2635, 0.6408, 0.3048

The table shows the equalizer coefficient vector that gives the minimum performance function achievable for different delay levels. Although delay 1, 2 and 3 gives zero error at corresponding cost functions, the most effective of them all is the delay that gives the minimum cost of 30.2020.

Table 2: Decision-Directed Equalizer

Initialization Spike	*Step size*	*Coefficient Vector(f)*	*Received Signal Vector(r)*	*Error(e)*
[0010]	0.012	-0.0927,0.1975, 0.6028, -0.0609	-1.100, -0.100, 0.900, 2.100	0.5453
[0010]	0.005	-0.0536,0.1138, 0.6568, -0.0920	-0.900, -0.900, -0.900, -0.900	-0.4447
[0010]	0.01	-0.0011, 0.1109, 0.6166, -0.1628	-2.100, -0.100, 2.100, 0.100	-0.3176
[0100]	0.01	0.1294 ,0.6749, -0.0604, -0.1628	-0.900, -0.900, -0.900, -0.900	-0.3872

Table 2 portrayed the error level and the coefficient or weight adjustment that compensated for the error at the corresponding step size to improve the output of the equalizer using decision directed equalizer. The best result was obtained when the error was reduced to -0.3176 at a step size level of 0.01

The convergence graph for the decision-directed equalizer at five levels of step-size values is shown in figure 6.

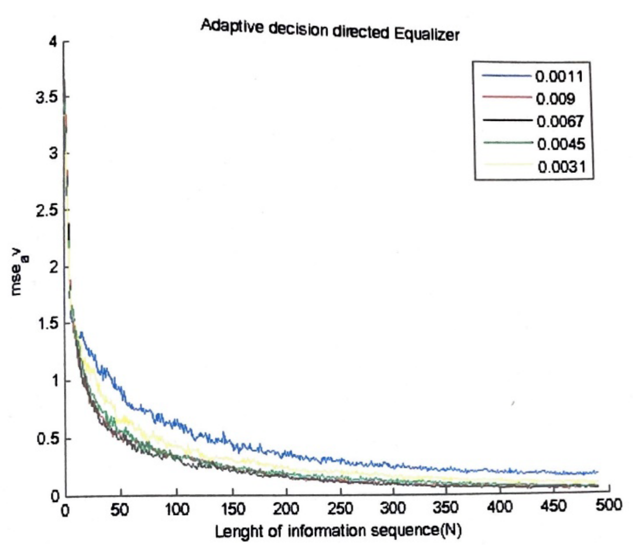

Figure 6: Equalization using Decision Directed LMS Algorithm

The signal waveform with step-size 0.0011,0.0031 and 0.0045 was able to maintain a constant amplitude of convergence from 250 points of the length of the sequence. Signals with step-size 0.0067 and 0.009 were able to converge at around 100 points of iteration. The optimal result was achieved at a step-size value of 0.009 followed by 0.0067. The error was minimized as shown by the ripple reduction in figure 6 and the convergence is almost abrupt at 50 iterations.

Table 3 showed the error level and weight adjustment that improves the output of the equalizer using the dispersion minimization equalizer.

Table 3: Dispersion Minimization Equalizer

Initialization Spike	*Step size*	*Coefficient Vector(f)*	*Received Signal Vector(r)*	*Error(e)*
[0010]	0.012	0.0843, -0.2411, 0.6158, 0.2887	-0.900, -0.900, -0.900, 2.100	0.0483
[0010]	0.005	0.0910, -0.2582, 0.5949, 0.2945	1.100, -1.100, -0.100, 0.900	0.3848
[0010]	0.01	0.0823, -0.2592, 0.5876, 0.2662	-2.100, -0.100, 2.100, 0.100	-0.1447
[0100]	0.01	-0.2285, 0.6117, 0.2872, 0.1632	-0.900, -0.900, -0.900, -0.900	-0.5975

The best result was obtained when the error was reduced to 0.0483 at a step size level of 0.012 for the dispersion minimization equalizer.

The convergence graph for the dispersion minimization equalizer at five levels of step-size values is shown in figure 7.
The optimal result was achieved at a step size value of 0.009 followed by 0.0067 because of the fast convergence of the graph.

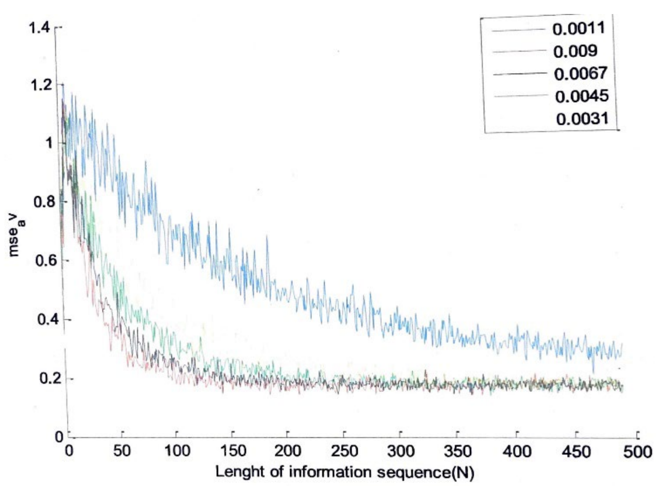

Figure 7: Equalization using Dispersion Minimization Algorithm

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A larger step size resulted in faster convergence and larger minimum mean square error (MSE) for both decision-directed and dispersion minimization algorithms. Decision-directed equalizer has a very fast convergence speed and scored optimal values that are within the MSE value of 10^{-2} as compared to dispersion minimization equalizer.

Conclusion

The destructive effects of multipath signal propagation on wireless radio signals were minimized using an adaptive decision-directed equalizer. The adaptive decision-directed equalizer has a very fast convergence speed and a larger minimum square error. Hence it was able to reduce the communication errors such as inter-symbol interference. Future research on this work should focus on the empirical application of equalization techniques using digital signal processing and the application of a genetic algorithm for channel equalization.

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