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Reduction of Destructive Multipath Signal Effects in Radio Propagation Using Adaptive Equalization Technique

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Abstract

Radio signals transmitted through propagation channels are subject to fading, dispersion, and distortion, often resulting in communication errors such as inter-symbol interference. These impairments, largely influenced by atmospheric conditions and physical obstructions, alter signal behavior during transmission. This study evaluates the effectiveness of adaptive equalization techniques in mitigating the adverse effects of destructive multipath signals in wireless and radio communication systems. A Rayleigh fading channel model was employed, and an adaptive algorithm was implemented to approximate the desired filter by minimizing the least mean square error (MSE) of the output signal. System performance was assessed using source recovery error as a cost function to correct delays and recover the transmitted information. Two equalization strategies—decision-directed and dispersion minimization algorithms—were developed to reduce multipath-induced errors. Results showed that adaptive equalization significantly reduced signal ripple. The decision-directed equalizer exhibited faster convergence but higher steady-state error, achieving an optimal MSE value of 10^{-2} and outperforming the dispersion minimization approach. These findings confirm the effectiveness of adaptive equalization in enhancing communication reliability under fading channel conditions.

Keywords: Destructive Multipath Signal; Fading Channel; Radio Propagation Channel; Adaptive Equalization; Least Mean Squares Algorithm

INTRODUCTION

The importance of radio communication has been proven to be a vital part of our daily activities. Additionally, the need for the proper functioning of radio communication systems to ensure accurate and reliable information transfer in radio propagation channels must be considered. In a radio communication environment, signals are transmitted and reach the receiver via a Direct Line of Sight (DLOS) or multiple paths. The primary causes of multipath propagation include reflection, refraction, diffraction, and scattering, which are caused by surrounding objects such as buildings, signposts, shrubs, trees, and mountains. Multipath propagation affects the transmitted signal, resulting in distortions and significant fading when it arrives at the receiver. The components of multipath signals are the delayed segments of the original wave that traveled through different paths, each with varying magnitudes, directions, and arrival times at the receiver. Since each component carries the original information, if the magnitude and phase (arrival time) of each component are measured at the receiver through a process called channel estimation, then all the multipath components can be combined coherently to enhance the reliability of information (Jina, 2023).

Fading is a phenomenon that occurs as a result of sudden and spontaneous variation in signal strength across any propagation medium (Yahong *et al.,* 2020). It is considered as a common phenomenon arising from signal characteristics variation in multipath propagation channels and therefore making the quality of received signal to be poor (Osuagwu, Nwabueze, & Muoghalu, 2024). In wireless communication systems, fading occurs due to multipath propagation and is sometimes referred to as multipath-induced fading. In wireless telecommunications, multipath is the propagation phenomenon that results in radio signals reaching the receiving antenna through more than one path. The effects of multipath at the receiver include constructive interference, destructive interference, and phase shifting of the signal. The distortion of signals caused by multipath is called fading (Mahdin & Sittul, 2017), which is usually associated with Rayleigh channel as evaluated for multiple input multiple output (MIMO) system in Achebe and Muoghalu (2024).

Generally, a signal transmitted in a channel experiences two types of fading: (1) large-scale fading and (2) small-scale fading. The mechanisms behind these two types of fading are diverse, but can be attributed to reflection, diffraction, and scattering (William, 2016). Other transmission impairments include free space loss, thermal noise, and atmospheric absorption.

The transmission path may also be corrupted by additive interference, such as that caused by other users. These noise components are usually presumed to be uncorrelated with the source sequence, and they may be broadband or narrowband, in-band or out of band relative to the band-limited spectrum of the source signal. Like the multipath channel interference, they cannot be known to the system designer in advance. Therefore, a technique has to be used to effectively cancel this impairment to transmitted signal. An example of such technique is the adaptive equalization method based on neural network model implemented in Osuagwu, Nwabueze, Muoghalu, and Ajere (2025). The equalizer has to reject such additive narrowband interferers by designing an appropriate linear notch filter based on the received signal.

When there is no Inter Symbol Interference (from multipath channel or from imperfect pulse shaping or from imperfect timing), the impulse response of the system from the source to the receiver has a single non-zero term. The amplitude of this single spike depends on the transmission losses, and the delay is determined by the transmission time. When there is inter-symbol interference caused by the channel, this single spike is scattered and duplicated once for each path in the channel. The number of non-zero terms in the impulse response increases. The channel can be modeled as a finite impulse response linear filter, and the delay spread is the total time interval during which reflections with significant energy arrive. The idea of an equalizer is to build a filter in the receiver that corrects these effects of the channel. In this work, equalization is performed so that the impulse response of the combined channel and equalizer has a single spike (Garima & Amandeep, 2021).

The objective of equalizer design lies in mitigating the effects of the channel and removing interferences. Therefore, the objectives of the research are as follows:

i. To characterize a digital communication channel (Nigerian Radio Space)

ii. To define the performance function using the matrix form of the “source recovery erroR”.

iii. To determine the least square cost function that will minimize delayed “source erroR “in equalizer design.

iv. To develop algorithms that will implement Adaptive Equalization.

v. To stimulate the Adaptive Equalization Using MATLAB/Simulink

vi. To find out the relationship between Adaptive approaches to trained equalizers and the blind equalizer algorithms, such as decision-directed and dispersion minimization.

METHODS

Working Principle

The structure shown in Figure 1 is an Adaptive equalization is a single-channel technique that is frequently used in receivers to compensate for linear distortion (ISI) introduced by the channel and transmit-receive filters (Richard et al., 2021). Given a channel of unknown impulse response, the purpose of an adaptive equalizer is to operate on the channel output such that the cascade connection of the channel and the equalizer provides an approximation to an ideal transmission medium (Prachi et al., 2024). The signal s(n) passes through a channel, and at the receiver end received signal (x(n)) is fed into the input of an adaptive filter, which tries to reverse the effect of the received signal by using filter coefficients. The output (y(n)) of the adaptive filter is then applied at the input of the decision device and also summed with the delayed version (d(n)) of the signal and fed back into the adaptive filter. An adaptive filter optimizes the received signal by best matching the filter coefficients. Adaptive equalizer changes the weights of the equalizer and enables it to adjust itself according to the modification in the channel. For this equalizer knows the transmitted data and it compares it with the received signal and evaluates the error, and finally adjusts it till the error is eliminated.

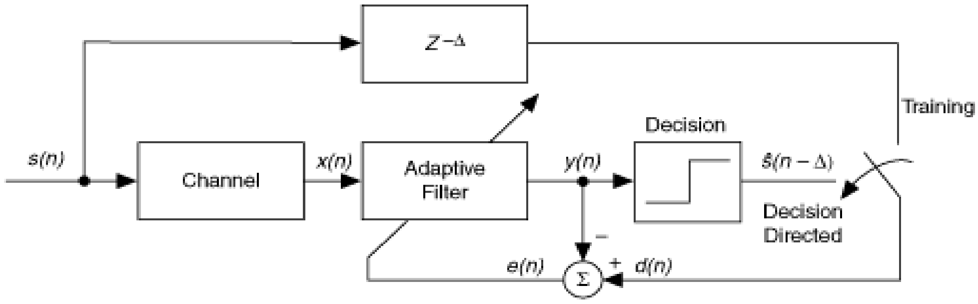

Figure 1: Adaptive Equalizer Block Diagram

Development of a Fading Channel Model

The model assumes that N equal strength rays arrive at a moving receiver with uniformly distributed arrival angles αn, such that ray nn experiences a Doppler shift $\omega d = \omega m \cos\alpha_n$ where $\omega m = 2\pi fv/c$ is the maximum Doppler frequency shift, v is the vehicle speed, f is the carrier frequency and c is the speed of light. Using $\alpha n = 2\pi fv/c - \pi/N$ *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, βn 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=100k*mph*, the carrier frequency f_c = 1800*MHz* for a typical GSM system, sample frequency fs = 10$k$bps, 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.

Adaptive Equalization Algorithms for Optimization of the Tap Coefficients

1. ***Least Mean Squares Algorithm (LMS)***

Least mean squares (LMS) algorithms are a class of adaptive filters used to mimic a desired filter by finding the filter coefficients that relate to producing the least mean squares of the error signal (difference between the desired and the actual signal) (Garima &

Amandeep, 2021). It is a stochastic gradient descent method in that the filter is only adapted based on the error at the current time. LMS filter is built around a transversal (i.e., tapped delay line) structure. Two practical features, simple to design yet highly effective in performance, have made it highly popular in various applications. LMS filter employs small-step-size statistical theory, which provides a fairly accurate description of the transient behaviour.

The output equation is defined as

$[k] = [k].[k]$ (2)

Where $[k]$ is the filter coefficient or weight vector, while $[k]$ is the input of the signal vector. The error signal $[k]$is the difference between the desired and actual signal and is defined as;

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

Where $[k]$ is the desired signal

The equation for optimizing the tap coefficients is

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

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, $\mu\mu$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

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

P_R is the received signal plus noise power

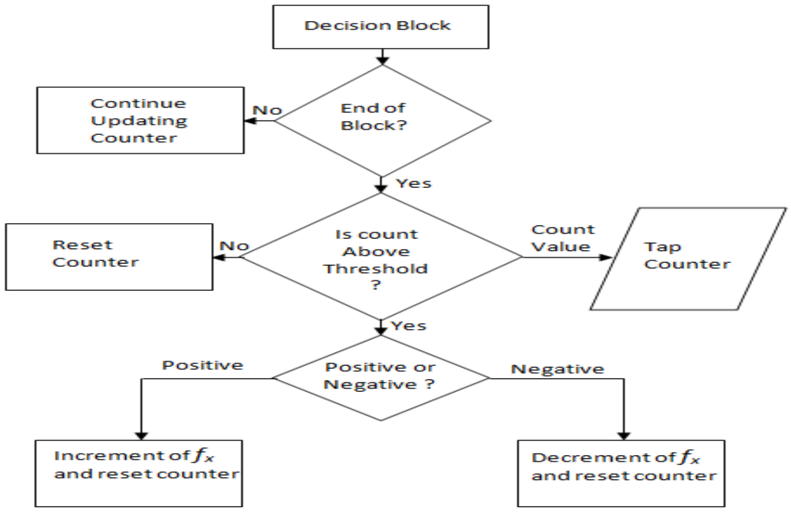

Figure 2: Coefficient Update Criteria for LMS

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;

$$e[k] = s[k-\partial] - y[k] \quad (5)$$

where $[k-\partial]$ 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;

$$J_{LS} = E^T E = (S - RF)^T (S - RF)$$

$$J_{LS} = S^T[1\text{-}R\,((R^T R)^{-1} R^T]S \quad (6)$$

Development of a 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.

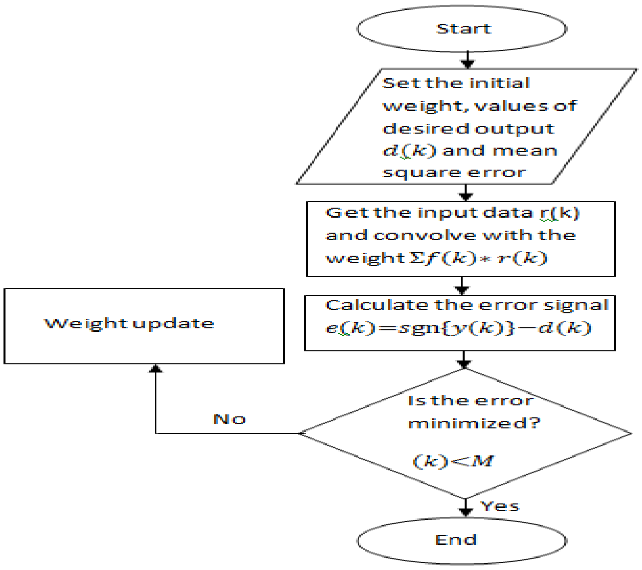

Figure 3: Decision-Directed LMS Algorithm

The weights are initialized, the error level is set, and the desired outputs are set. The weight multiplies the sampled number of time-delayed versions of an incoming input signal, which 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.

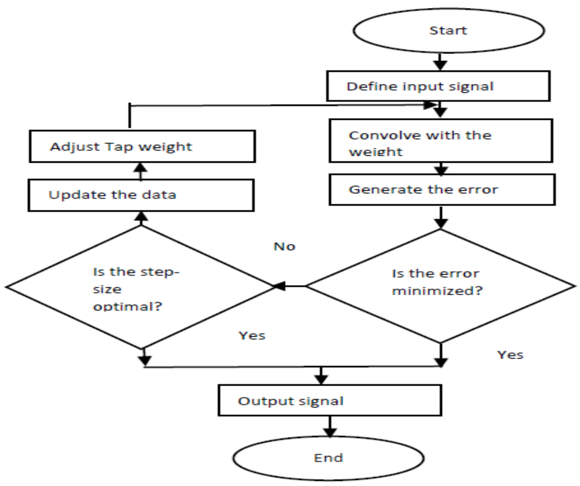

Figure 4: Dispersion Minimization Equalization Algorithm

RESULTS AND DISCUSSION

Analysis of Received Signal Strengths

Field measurements taken at Awka by the researcher, results are recorded for three locations where the measurements are performed. The locations are (i) the point at the front of Sports Club Amawbia Location I, $6^0$11'43.2"N, $7^0$03'31.1"E, (ii) at College Demonstration Nursery/Primary School Awka Location II, 06^0 14'47.2"N, $007^0$05'58.8"E and (iii) Saint Patrick Cathedral Gate Awka Location III**,** 06^0 13'18.9"N, $007^0$04'28.9"E. The results obtained and listed in Tables 1 to 3 show the received signal strength from different base transceiver stations as seen from the test points.

Table 1. Field measurement taken at location I

BTS Name	Sector Name	Distance From BTS (Km)	Test point (bearing)	RSS at Test Point (dBm)	Received power (μw)	Antenna Height (m)	Freq. of Operation (MHz)
AN0012	Sector-2	0.21	$6^0$11'43.2"N $7^0$03'31.1"E	-69	0.13	24	1800MHz
AN0012	Sector-3	0.21	"	-71	0.079	"	"
AN0031	Sector-1	3.7	"	-62	0.63	"	"
AN0004	Sector-1	2.57	"	-72	0.063	"	"
AN0028	Sector-3	1.47	"	-80	0.01	"	"
AN0072	Sector-2	1.45	"	-77	0.019	"	"
AN0033	Sector-2	1.48	"	-69	0.13	"	"

Normal BTS power is-35db, service efficiency is-44 To-65dB.

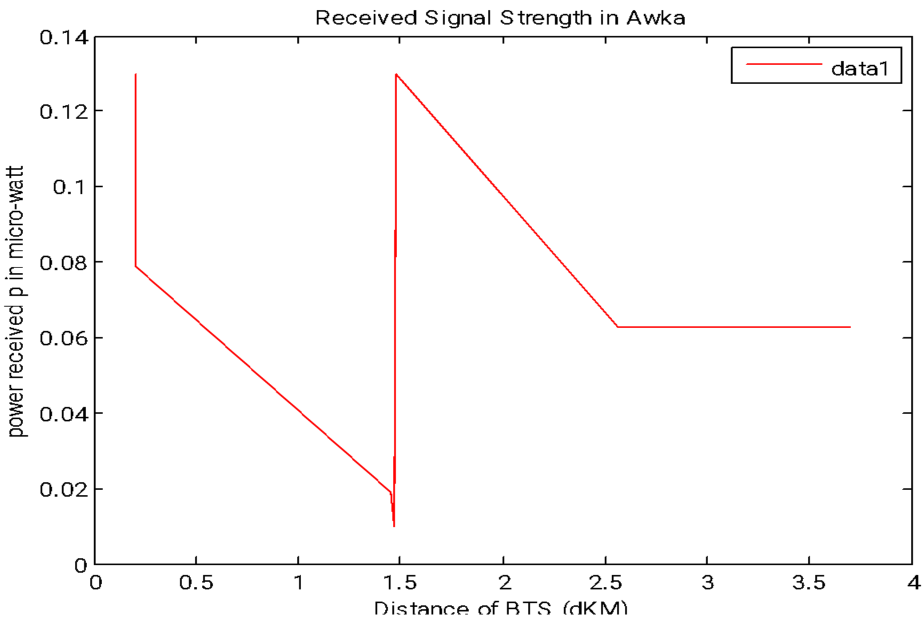

Figure 5: Representation of RSS at Location I

Table 2. Field measurement taken at location II

BTS Name	Sector Name	Distance From BTS (Km)	Test point	RSS at Test Point (dBm)	Received power (μw)	Antenna Height (m)	Freq. of Operation(MHz)
AN0114	Sector-2	0.61	06^{0} 14’47.2”N 007^{0}05’58.8” E	-68dBm	0.158	24	1800MHz
AN0114	Sector-1	0.61	‘	-81dBm	0.0079	‘	‘
AN0014	Sector-1	2.15	‘			‘	‘
AN0014	Sector-3	2.15	‘	-74dBm	0.040	‘	‘
*AN0451	Sector-3	17.72	‘	-71dBm	0.079	‘	‘
*AN0451	Sector-1	17.72	‘	-74dBm	0.040	‘	‘
AN0034	Sector-1	1.61	‘	-79dBm	0.0125	‘	‘
@AN0381	Sector-3	4.90	‘	-85dBm	0.0031	‘	‘

*This station, asterisked, is located far away at Igbariam Junction by expressway. The other base Station marked with @ is situated at Ukpo Junction by Express way

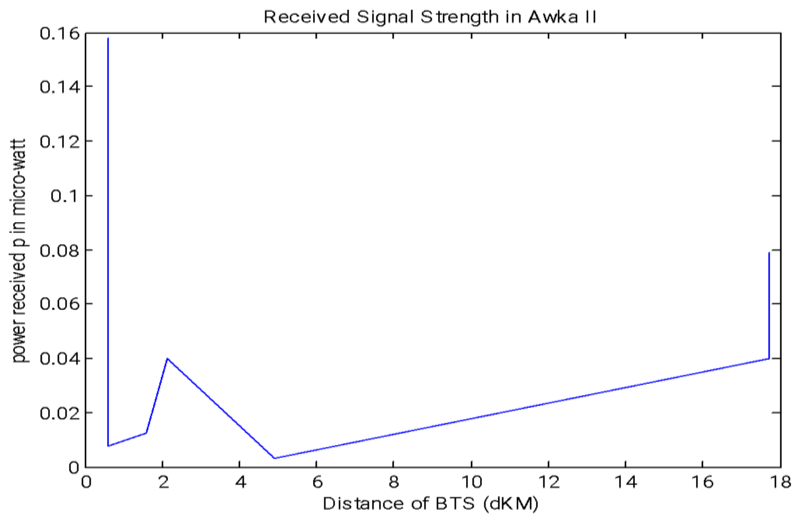

Figure 6: Representation of RSS at Location II

Table 3. Field measurement taken at location II

BTS Name	Sector Name	Distance From BTS (Km)	Test point	RSS at Test Point (dBm)	Received power (μw)	Antenna Height (m)	Freq. of operation (MHz)
AN0113	Sector-3	10.38	06^{0} 13’18.9”N 007^{0}04’28.9” E	-77dBm	0.019	24	1800MHz
AN0033	Sector-3	2.14	‘	-82dBm	0.0063	‘	‘
AN0031	Sector-1	7.27	‘	-85dBm	0.0031	‘	‘
AN0072	Sector-1	4.17	‘	-86dBm	0.0025	‘	‘
AN0029	Sector-2	1.1	‘	-89dBm	0.0012	‘	‘
*AN0307	Sector -2	10.66	‘	-85dBm	0.0031	‘	‘
@AN0308	Sector-2	8.52	‘	-85dBm	0.0031	‘	‘

The number of base transceiver stations in Awka town is twelve (12 numbers).

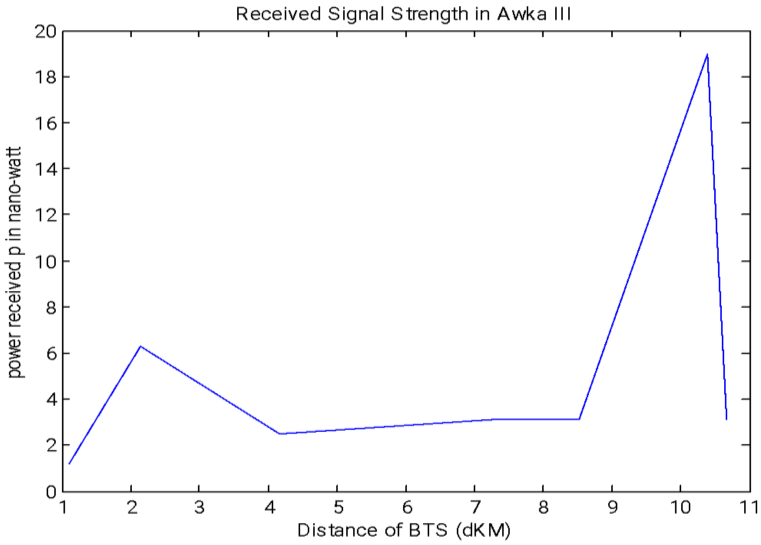

Figure 7: Representation of RSS at location III

At location II, five base transceiver stations are identified as itemized in Table 2. At BTS-AN0114, the signals from two diversity antenna sectors are picked by the instrument, showing their signal strength as-68dBm for sector 2 and-81dBm for sector 1. From this variation of the signal strength received at a point, it can be deduced that the two signals arrived at the receiver through two different paths. Although the two signals seem to arrive at the receiver through the same length of Transmitter/Receiver separating distance, one of the received signals might have taken a longer time to arrive due to obstruction or any other diversity effect. The same effect is noticed at the signals received from BTS-AN0451, which showed that sector-3 has RSS of-71dBm while sector-1 has RSS of-74d Bm. The same diversity effect is noticed at location-I (Table 1) where sector-2 of AN0012 and sector-3 of AN0012 produced received signal strength of-69dBm and-71dBm, respectively, with Transmitter/Receiver separation distance of 0.21 Km.

Determination of Source Recovery Error

A delay in any system has a serious effect on the output performance. It is normally expected that what is transmitted should be religiously received as a whole at the receiver without impurity. But due to a delay in the process, the transmitter gives out a delayed /attenuated/smeared version of what is transmitted, and that is the cause of the error.

The performance function is a model that tries to correct the effect of the delay, thereby reducing the error to recover the original source information.

The result of the simulation of Equation (7) are stated for the least square performance function as Table 4.

$$J_{LS} = S^T S - 2S^T RF + (RF)^T RF \quad (7)$$

Table 4. Minimum performance function

Delta	Error	J_{min}	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

Table 4 shows the equalizer coefficient vector that gives the minimum performance function achievable for different delay levels. Although delays 1, 2, and 3 give zero error at corresponding cost functions, the most effective of them all is the delay that gives the minimum cost of 30.2020.

Determining the MSE Using Decision Directed Equalization Approach

Result of the simulation obtained when the initializatioN 'spikE 'is located at [0 0 1 0].

Table 5. Decision Directed Equalizer

Initialization Spike	Step-size(μ)	Coefficient vector (f)	Received signal vector(r)	Error (e)
[0010]	0.012	-0.0927,0.1975,0.6028,-0609	-1.100,-0.100,0.900,2.100	0.5453
'	0.005	-0.0536,0.1138,0.6568,-0.0920	-0.900,-0.900,-0.900,-0.900	-0.4447
'	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 5 portrays the error level and the weight adjustment that compensates for the error at the corresponding delay to improve the output of the equalizer. The best result is obtained when the error is reduced to-0.3176 at a step-size of 0.01 for the Decision Directed equalizer.

Figure 8 is the Convergence graph for the Decision Directed equalizer at three levels of step-size values. The optimal result is achieved at a step-size value of 0.009, followed by 0.0067. The graph shows that the decision-directed equalizer has a very fast convergence speed as compared to trained LMS and Dispersion Minimization algorithms. It can be seen that the error is minimized as shown by the ripple reduction in the graph of Figure 8. The convergence is almost abrupt at about 10 iterations.

$$f_i[k+1] = f_i[k] + \mu(sign(y[k]) - y[k])r[k-i] \quad (8)$$

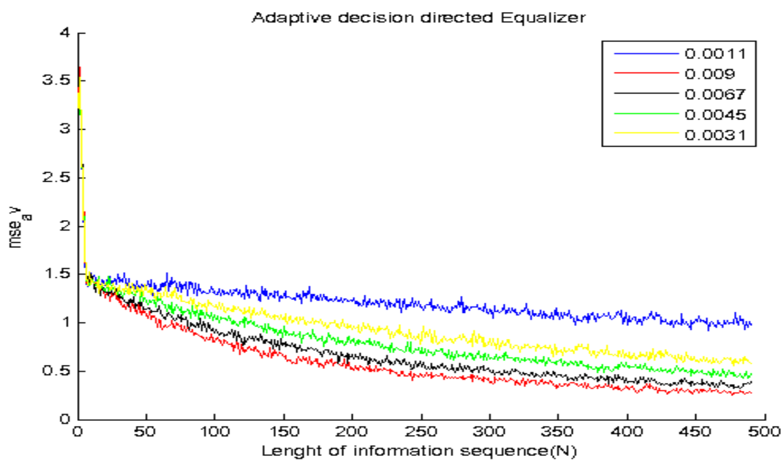

Figure 8: Equalization using the decision directed LMS algorithm

Determining the MSE Using Dispersion Minimization Equalization Approach

Result of simulation obtained when the initializatioN ‘spikE ‘is located at [0 0 1 0].

Table 6. Dispersion Minimization Equalizer

Initialization Spike	Step-size(μ)	Coefficient vector (f)	Received signal(rr)	Error (e)
[0010]	0.012	0.0843,-0.2411,0.6158, 0.2887	-0.900,-0.900,-0.900, 2.100	0.0483
‘	0.005	0.0910,-0.2582,0.5949, 0.2945	1.100,-1.100,-0.100, 0.900	0.3848
‘	0.01	0.0823,-0.2592,0.5876, 0.2662	0.100,-2.100,-0.100, 2.100	- 0.1447
[0100]	0.01	-0.2285,0.6117,0.2872, 0.1632	-0.100,2.100,0.100,-0,900	- 0.5975

Table 6 portrays the error level and the weight adjustment that compensates the error at the corresponding delay to improve the output of the equalizer. The best result is obtained when the error is reduced to 0.0483 at a step-size of 0.012 for the Dispersion Minimization equalizer.

From the above results, it is deduced that the error is reduced optimally wheN 'spikE 'is located at position [0 1 0 0] for a step size of 0.01 compared to thE 'spikE 'location of [0 0 1 0] for a step size of 0.01.

Figure 9 is the Convergence graph of Dispersion Minimization at five levels of step-size values. The optimal result is achieved at a step-size value of 0.009 followed by 0.0067 because of fast convergence of the graph.

$$f_i[k+1] = f_i[k] + \mu(1 - y^2[k])y[k]r[k-i] \quad (9)$$

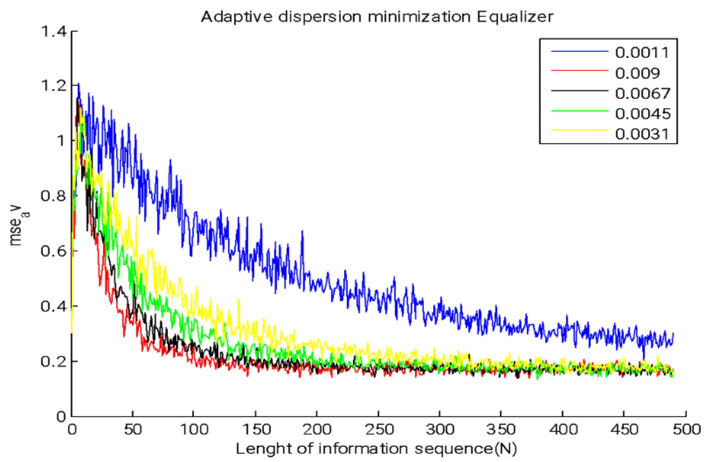

Figure 9: Equalization using dispersion minimization algorithm

Determination of Adaptive Equalizer Convergence

The graph of Figure 9 represents an adaptive channel equalization for system identification in which a predefined signal is passed through with added white Gaussian noise. The ripples are more pronounced showing the presence of additive noise and fast fading. LMS algorithm is used to minimize the mean square error that determines the inverse of the channel output, which gives the same result as the input.

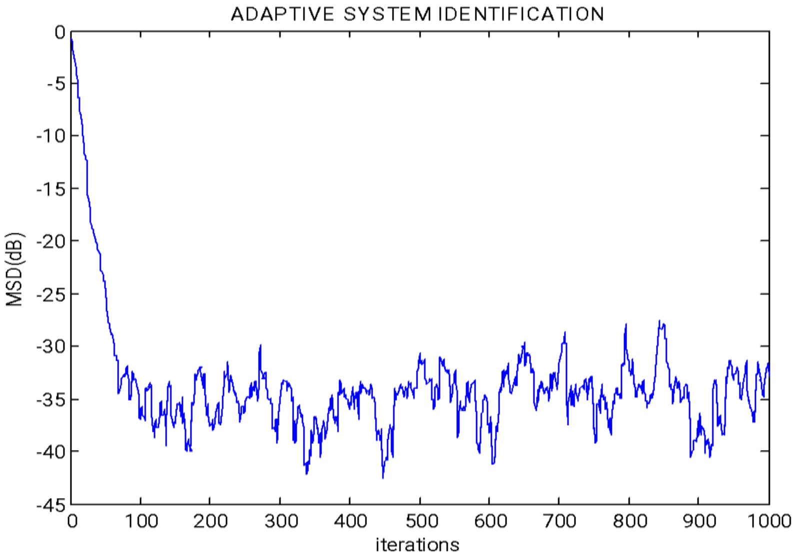

Figure 9: Adaptive channel equalization at a step size $\Delta = 0.5$

CONCLUSION

This work surveyed the different components of the communication system. The Nigerian Radio Space characterization has been surveyed. On the characterization of the Nigerian Radio Space, a conclusion is drawn from data obtained from the various sample surveys, which confirmed that Nigerian Radio Space Refractivity differs from the international telecommunication standard specification recommendation (ITU-R 853). The result shows that the refractivity of Nsukka, South East Nigeria, differs by 13.05% from that recommended by ITU-R for Nigerian radio Communication Space.

Also, the received signal strength of the Airtel GSM Communication base stations is also surveyed from three locations in Awka, Anambra State. The signals thus received are expected to be filtered and equalized in order to regain the original copy of the transmitted signal. Lack of adequate equipment stalled further empirical work on equalization; hence, the use of MATLAB to simulate in order to realize an output that is devoid of signal interference and noise. The equalization is done using the Least Mean Square Algorithm for the linear equalizer. The Techniques applied at this work include: the Source Recovery Least Squares technique, the Trained Linear Least Mean Square algorithm, the Decision Directed algorithm and the Dispersion Minimization Algorithm. The relationship governing the three techniques differs in their error terms. In the case of the training equalizer, the error term, which is the difference between the source (transmit) symbol and the equalizer output signal, is used. In the decision-directed technique, the delayed source signal is not used; instead, the error is now computed directly using the signal at the output of the detector/decision device as the desired output, which is compared with the actual output of the equalizer. While in the dispersion minimization algorithm, the error is the deviation of the equalizer output from the squared source-gamma. The result of the finding shows that optimum step-size is 0.045 at mean square error-average (MSE_av) was less than 10^{-2}, the filtration capability of the adaptive equalizer is dependent on the variable factors such as equalizer coefficient, step-size, and type of algorithm used to minimize the error, convergence time of the adaptive equalizer is a function of such variable factors as step-size, equalizer coefficients, and the type of algorithm used, and optimum value of the adaptive equalizer design involves variable parameters with effective tracking ability to the frequency changes of the input signal interference.

REFERENCES

Achebe, P. N. and Muoghalu, C. N. (2024). Capacity analysis of multiple antenna wireless system under different array and channel configurations. *International Journal of Advanced Networking and Applications, 15*(4), 6063-6068.

Garima, M. and Amandeep, S.S. (2021). Adaptive equalization algorithm: An overview. *International Journal of Advanced Computer Science and Applications* vol. 2, No.3.

Jina, K. (2023). Area and power conscious rake receiver design for third generation WCDMA systemS'.

Mahdin, M. and Sittul, M. (2017). Comparison of different models for the analysis of rayleigh fading channels.

Osuagwu, I. C., Nwabueze, C. A., & Muoghalu, C. N. (2024). Investigating the quality of service at receiver end due to multipath fading on signal strength. *International Journal of Progressive Research in Engineering Management and Science, 4*(6), 2552-2557.

Osuagwu, I. C., Nwabueze, C. A., & Muoghalu, C. N., Ajere, I. U. (2025). Adaptive equalization technique for mitigating effect of multi terrain multipath signal quality. *International Journal of Advanced Networking and Applications, 16*(5), 6533-6543.

Prachi, S., Piush, G. and Pradeep, K. S. (2024). Performance comparison of ZF, LMS and RLS algorithms for linear adaptive equalizer. *Advance in Electronic and Electric Engineering.* ISSN 2231-1297, 587-592.

Richard, C., North, R. A., and James, R. Z. (2021). The performance of adaptive equalization for digital communication systems corrupted by interference. *ASILOMAR Conf. on Signals, Systems and Computers10/93, pp 3.*

William, C. L. (2016). Mobile communication engineering, McGraw-Hill book company, New York. www.navymars.org/national/training/nmo.14182_ch2.pdf, "Radio Wave PropagatioN".

Yahong, R., Zheng, J., and Jing, P. (2020). Simulation model with correct statistical properties of rayleigh fading channels