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Enhancement of FBMC Communication through Equalisation Techniques

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Abstract

As the standarisation of Communication techniques for 5G are getting discussed all over the world, Filter Bank Multi-carrier Communication (FBMC) stands in the front as the leading contender. While the FBMC has more spectrum efficiency, more flexibility to use different pulse shape and efficient power control, this FBMC System can also be susceptible to Synchronisation Errors, Time Offset and Frequency Offset. Though ICI (Interference of Symbols from other subcarriers in the same index) and ISI (Interference of Symbols in different time index from all subcarriers) depends upon Pulse-shape, Channel and Data, it can be mitigated through Channel Equalisation, Channel Estimation and Synchronisation techniques. In this paper, we explore different Equalisation techniques such as Adaptive Equalisation [Zero Forcing (ZF), Mean Square Error (MSE) Criteria (Least Squares Method and Least Mean Square (LMS) Algorithm)] and Non-Linear Equalisation Techniques [Decision Feedback Equalisation (DFE), Maximum Likelihood (ML) criteria and Minimum Mean Square Error (MMSE) Techniques]. We have proposed a novel approach in Decision Forward Equalisation (DFE) techniques with the best iterative Minimum Mean Square Error (MMSE) Algorithm on FBMC Communication and compared the results.

Keywords: SDR, OFDM, FBMC, ISI, ICI, Equalisation, MSE, DFE.

INTRODUCTION

As the fourth generation telecommunication is entering next generation, all the Radio communications are becoming software oriented. Software Defined Radio (SDR) is initially addressed followed by the technology of Filter Bank Multicarrier Communication (FBMC). The FBMC techniques are prone to ISI and ICI. The mitigation techniques of Equalisations are discussed. Results of different Equaliser with noise are compared. In this paper a novel DFE techniques with best iterative MMSE is proposed for FBMC and implemented and the results are given.

As the modern telecommunication is improving over each decade, two trends continue to persist. They are data rate and ability to send information independent of location and time. The software based radio communication enables the latest ICT devices a reconfigurable one. Data rate demand is increasing beyond 1 Gbps. The present 4G LTE [1] technology uses OFDM communication techniques. This has inherent problem of spectrum leakage and maintenance of Orthogonality among all sub-carriers. There is need for high data rate and low latency, as the world moves towards fifth generation Communication. Filter Bank Multicarrier Communication (FBMC) is very old FDM technology and due to complexity in implementation, this could not see the standards. With the advent of new fast processors, FBMC is suggested as next generation Communication layer and is implemented [2]. This on real world atmosphere will be prone to ISI and ICI. In order to mitigate this through Equalisation, in Section-II concept of Software Defined Radio (SDR) and working of FBMC is described. Then Section-III deals with the concept of Equalisation techniques and its implementation. In Section – IV a comparison is brought out with various FBMC Equalisers. Section-V gives the conclusion and future research work.

CONCEPTS OF SDR AND FBMC

A. SDR[2][3]

The SDR Forum which is working alongwith IEEE P1900.1 group has defined SDR as “Radio in which some or the entire physical layer functions are software defined”. Radio is a device which transmits and / or receives signals in the Electro-magnetic Spectrum to transfer information such as voice, music, and data. Scientifically the frequencies are grouped from 3 KHz to 300 GHz as Extremely Low Frequency (ELF) to Extremely High Frequency (EHF) under the various frequency bands.

In SDR-Transmitter the information is formatted into Digital input, in the form of message symbols. Then these symbols

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could be converted into bit stream, *a[n]*, by Pilot Sequencing. After Pulse Shaping the digital Baseband signal m[k] is formed. Then the Digital signal is converted into analog baseband signal, *m(t)*. After using modulation such as PAM, QAM, PSK, FSK, OFDM or its variants the Electromagnetic waves suitable to antenna characteristics are amplified and transmitted. After it travels thru' media, the signal is received at the receiver.

In SDR-Receiver, the received signal from Antenna is demodulated using appropriate Demodulation techniques such as OFDM, QAM, FSK, PSK and its variants. Symbols are recovered, after Frame Synchronisation, Pilot Sequencing, Frequency Synchronising and Channel Equalisation, *y[n]*. To make the bit stream decoding is carried out to recover the original the message successfully. The SDR Tx-Rx is given in the figure 1. below.

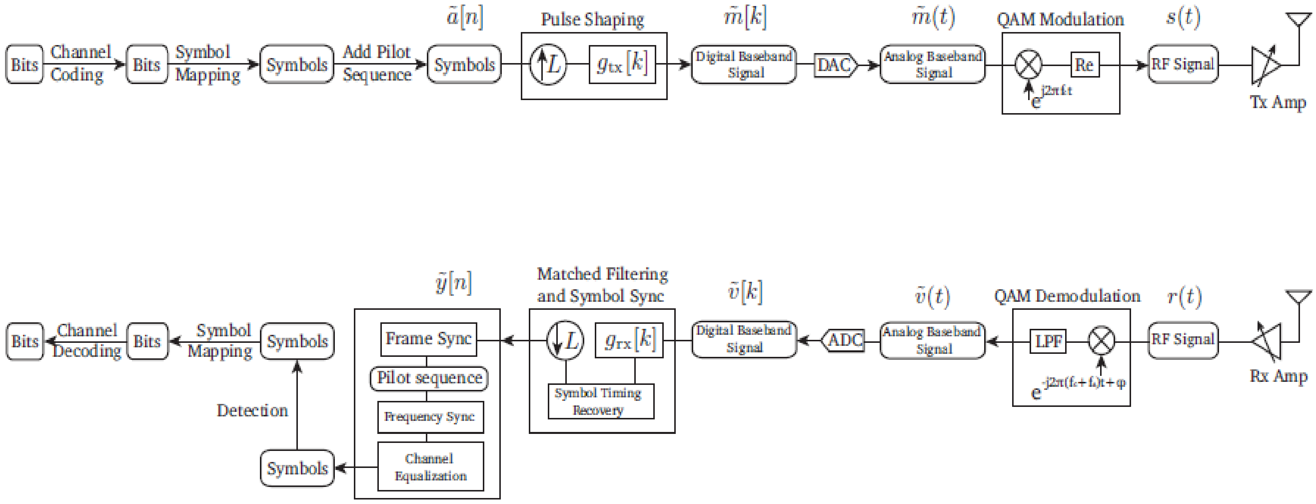

Figure 1: Block diagram of SDR Tx and Rx.

Most of the above functions can be executed in a Digital Signal Processor (DSP) or Field Programmable Gate Array (FPGA) or General Purpose processor (GPP) using Algorithms, middleware, Rate Convertor software. Pure Software Radio is yet a distant dream due to high architectural Complexity and high Flexibility. Software Defined Radio (SDR) is the one which has medium architectural Complexity and more than medium Flexibility. These radios are reconfigurable one where in the software can be upgraded or added with other modes of radio or frequency of operation.

B. FBMC [4] [5]

Digital Modulation modulates fundamentally three parameters viz., Amplitude (*A*), Phase (θ_k) and Frequency (f_c) of sinusoidal signal. Mathematically, to represent ASK, PSK and FSK, this can be written as,

$$s(t) = A \cdot \cos(2\pi \cdot f_c \cdot t + \theta_k) \tag{1}$$

Filter Bank Multi Carrier (FBMC) [5] [6] communication systems are a subclass of Multi-carrier Systems. This technique was first developed in the mid-1960s. Chang [7] and Saltzberg [8] have introduced PAM Symbol transmissions and QAM symbol Transmissions parallelly with filter-banks theory for efficient Bandwidth management. There are three major types of FBMC. They are suggested by different authors [6]. They are as follows:-

<u>FMT</u>. Filtered Multi Tone FBMC uses subcarrier bands with no overlap. Data symbols are QAM.

<u>CMT</u>. Cosine Modulated multi tone-FBMC uses subcarrier with maximum overlap.

<u>SMT</u>. Staggered Modulated Multi tone-FBMC uses subcarrier bands with maximum overlap. Data Symbols use PAM with VSB Modulation. It is also known as Offset QAM (OQAM).

FMT based FBMC is built on the conventional method known as Frequency Division Multiplexing (FDM).

FMT-FBMC System is a filter bank modulation technique in which, N branch filters are frequency-shifted-baseband filters, called prototype filter, that achieves high level of spectral containment. ICI is resolved through use of well-designed filters with high stop-band attenuation. ISI may be compensated for by adopting the conventional method of square-root Nyquist filtering. But the severe impediments due to Channel variations, ISI will persist and can be removed by equalisation techniques.

C. FBMC Technology

FBMC Communication uses Sample Rate Converters (UP and DOWN), Prototype Filter and Filter Banks as backbone element. The input-Output relation of Up Converter is given as

$$y_k[n] = \begin{cases} x[n/L], \; n = mL, \; m \text{ is an integer} \\ 0, \quad \text{otherwise} \end{cases} \tag{2}$$

The input-Output relation of Down Converter is given as

$$y_D[n] = x[Mn], \; M \text{ is an integer} \tag{3}$$

Prototype Filter is designed to match the time and frequency spread of channel which is given below:-

Time spread

$$\sigma_t = \sqrt{\int_{-\infty}^{\infty} t^2 |h(t)|^2 dt}$$

Frequency spread

$$\sigma_f = \sqrt{\int_{-\infty}^{\infty} f^2 |H(f)|^2 df}$$

Choose h(t) so that H(f) = h(lf), for a constant scaling factor l. Then one may find that $(T/\Delta\tau) = (F/\Delta\gamma)$

$$Or\ T/F = \Delta\tau/\Delta\gamma = \sigma t/\sigma f = l \tag{4}$$

Let $g(t) = e^{-\pi t^2}$ be the Gaussian pulse. The first property for the filter is $G(f) = g(f)$. The second property is the Heisenberg-Gabor uncertainty principle which states that for an arbitrary pulse $h(t)$, the equality $\sigma_t \,.\, \sigma_f \geq (1/4\pi)$ holds when $h(t) = g(t)$

Let p(t) be a prototype filter and with N sample points in the filter length βN + 1, where β is an integer greater than 1. Then β parameters are to be found for optimal design of prototype filter.p[n] is given as

$$= \begin{cases} \frac{1}{\beta N+1}\left(k_0 + 2\sum_{l=1}^{\beta-1} k_l \cos\left(\frac{2\pi l n}{\beta N+1}\right)\right), & 0 \leq n \leq \beta N, \\ 0, & \text{Otherwise.} \end{cases} \tag{5}$$

The best prototype filter, *p[n]*, designed as part of PHYDYAS Filter[9] at Europe is given below:-

The impulse response is given as

$$p[m] = 1 + 2\sum_{k=1}^{K-1} (-1)^k P_k \cos\left(\frac{2\pi k}{MK} m\right)$$
$$p[0] = 0, \tag{6}$$

The frequency response is given as

$$P(f) = \sum_{k=-(K-1)}^{K-1} P_k \frac{\sin\left(\pi\left(f - \frac{k}{MK}\right)MK\right)}{MK \sin\left(\pi\left(f - \frac{k}{MK}\right)\right)} \tag{7}$$

$$\begin{aligned} m &= 0, 1, \ldots, KM-2 \\ K &= 4 \\ \bar{P}[0] &= 1 \\ \bar{P}[1] &= 0.97195983 \\ \bar{P}[2] &= 1/\sqrt{2} \\ \bar{P}[3] &= 0.23514695 \end{aligned}$$

The above is pictorially given as follows:-

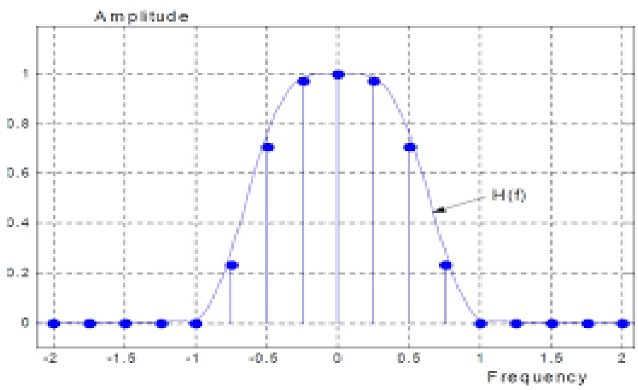

Figure 2: Frequency Response

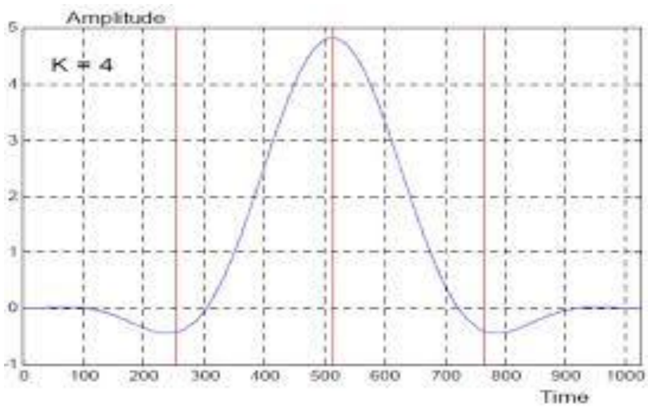

Figure 3: Impulse Response

The same was simulated through MATLAB and the results are depicted below:-

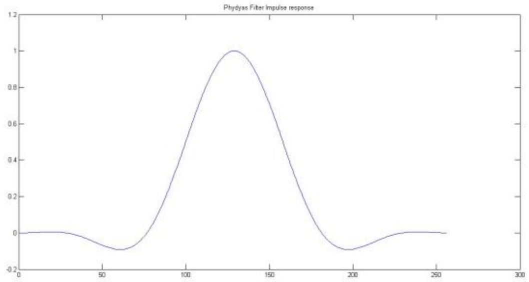

Figure 4: PHYDYAS Filter Impulse Response

Filter Banks

Filter Bank is a system that comprises of a Group of Filters which processes a common input or result into a common output. Filter banks either break down an input signal to form sub-band component signals or combine the sub-band signals to form the output signal. There are two major types of Filter banks viz., Analysis Filter Bank (AFB) and Synthesis Filter Bank (SFB). AFB is used for analyzing the input signal according to characteristics of each filter. SFB is used to filter the individual signals and added to get combined new composite signal. For harnessing the real power of filter banks, it has to be used in pairs, either AFB-SFB combination as Sub-band Systems or as SFB-AFB pair as Trans-multiplexers. SFB-AFB combination is used in multicarrier communication as Tx-Rx pair. Decimation, down conversion takes place at AFB and it consists of the filtering of the input signal (anti-aliasing Filtering) and subsequent down-sampling. Interpolation, i.e., takes place at SFB and it consists of an up-sampler and an interpolation filter (Anti-imaging filtering). Modulated Filter Banks are frequency shifted versions of Low Pass Prototype. It is achieved by multiplying the prototype with a modulation function. FMT-FBMC-Tx is implemented as follows with N number of Sub carriers, $s_k[n]$ data symol on the k[th] subcarrier FMT Symbol, L= KN, symbol period, K oversampling factor and $x[n]$ is the transmitted FMT- FBMC signal.

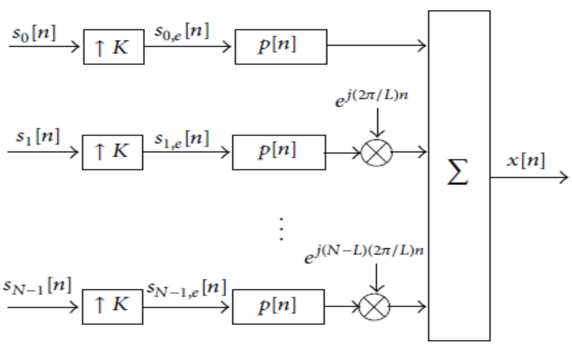

Figure 5: FBMC FMT Transmitter Block Diagram.

$$x[n] = \sum_{k=0}^{N-1} \left(\sum_{m} s_k[m]\, p[n-mK] \right) e^{j(2\pi kn/L)}$$

$$= \sum_{m} \left(\sum_{k=0}^{N-1} s_k[m]\, e^{j(2\pi kn/L)} \right) p[n-mK]. \tag{8}$$

FMT-FBMC-Receiver is implemented as follows:-

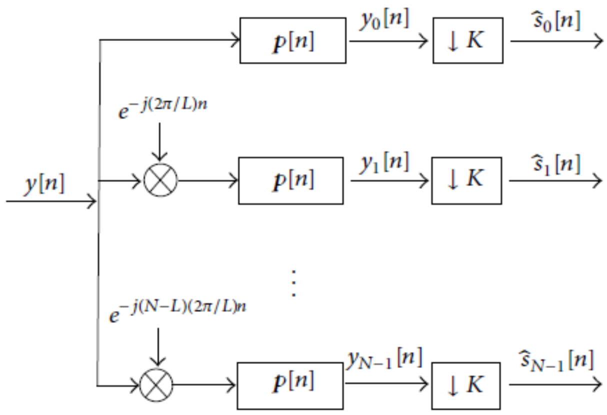

Figure 6: FBMC FMT Receiver Block Diagram.

$$y_k[n] = \left(y[n]\, e^{-j(2\pi kn/L)} \right) * p[n]$$

$$= \sum_{m} y[m]\, e^{-j(2\pi km/L)}\, p[n-m] \tag{9}$$

$$\hat{s}_k[n] = y_k[Kn] \tag{10}$$

With Polyphase network, FMT-FBMC system can be implemented with least complexity [4] [6].

Simplified FBMC Implementation and Results

A simple FBMC Tx-Rx is implemented by us using MATLAB. The block diagram of FBMC system is given in Figure.7.

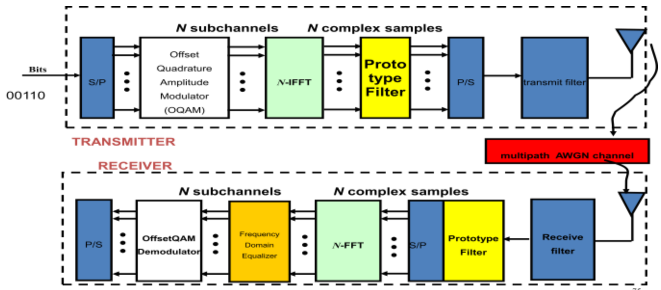

Figure 7: Block diagram of Simple FBMC System.

The constellation diagram of above FBMC with 4 QAM symbols, 64 Subcarriers, 4 Frames and AWGN Channel is given in figure.8.

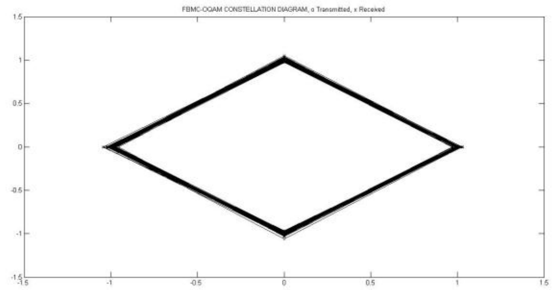

Figure 8: Constellation Diagram of 4-QAM- FBMC with AWGN Channel both transmitted and received symbols.

The Bit-Error-Rate for the above FBMC system in AWGN Channel is given at figure.9.

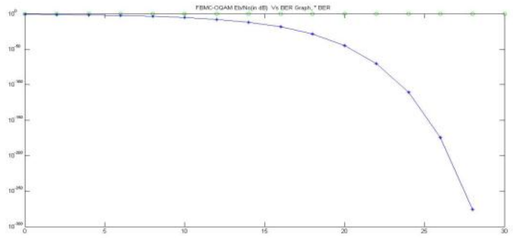

Figure 9: Bit-Error-Rate for FBMC in AWGN Channel

The FBMC Transmitted pulse power is depicted at figure.10.

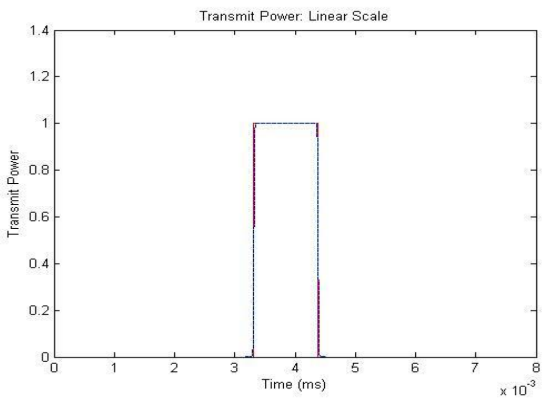

Figure 10: Transmitted Pulse of 4-QAM- FBMC.

CONCEPTS OF EQUALISATION [10][11][12]

Equalisation is an enhancement technique in Signal processing domain to combat Inter Symbol Interference (ISI). Every communication modulation technology is prone to ISI, due to channel variation over time. For high-speed data transmission, we have to use adaptive techniques of equalisation. The Equaliser functions like a delta channel response on a received signal. A simple block diagram of communication system with equaliser is given below:-

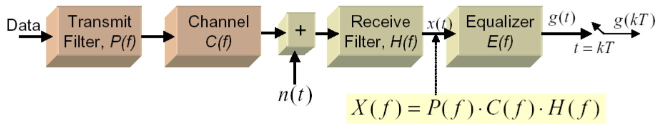

Figure 11: Basic Communication Block Diagram with Equaliser.

Emprically, the equaliser function in amplitude and phase can be represented as follows:-

$$|G_E(f)| = \frac{1}{|G_C(f)|} \Rightarrow |G_E(f)|\cdot|G_C(f)| = 1 \Rightarrow h_{total}(t) = \delta(t)$$
$$\arg(G_E(f)) = -\arg(G_C(f)) \qquad (11)$$

The equaliser is a kind of inverse filter of the channel. If the channel is frequency selective, the equaliser enhances the frequency components with small amplitudes and attenuates the frequency components with larger amplitudes so as to obtain a flat, frequency response with linear phase. The equalizers can be broadly classified into Linear and NonLinear type. Linear equalizers are of feed forward only type of filters. Non-linear Equalisers will have both feed-forward and feedback path, to take to change the subsequent outputs of the equalizers. In order to design an equaliser, there are two components to be considered viz., Structure and Algorithms. Filter structure could be either Tranversals or Lattice type. In Linear Tranversal Equalisers(LTE), the tap points delay by time(T) of Symbol Duration(S) and number of delay elements, n, would be the function of delay spread of the channel. The second type is Lattice equaliser, wherein, the signal is divided into two portions. One is forward loop and other is backward loop. The delay taps and reflection coefficients are used recursively to obtain the output. The lattice structured equaliser has advantages such as Numerical stability, Faster convergence and Dynamic Assignment. The disadvantage is its complex structure. The equaliser has to intelligently decide on the number of tap joints(Coefficients of filters), as it should be depending on the delay spread experienced in the channel, in which the communication devices are placed. The coefficients should change for converging the error(output-input) into minimum mean square error. In the non-linear category there are three methods such as Decision feedback Equalisation, Maximum Likelihood Symbol Detection(MLSD) and Maximum Likelihood Sequence Estimation(MLSE). Under Adaptive category(both Linear and Non-linear), there are three algorithms viz, Zero-Forcing(ZF), Least-Mean-Square(LMS) and Recursive Least-Square(RLS) Algorithm.

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There are four factors in adaptive equaliser which will decide the algorithms. They are Rate of Convergence to optimum solution, Mis-adjustment to optimal MSE, Computational complexity and Numerical Properties of inaccuracy calculations.

A. ZERO FORCING(ZF) EQULAISER

An analogue signal, *x(t)* is sampled at times, *t=nT* to give a digital signal $x_n=x(nT)$, n=0,1,….∞. The Z-Transform of x_n is defined analogously to the Laplace transform of a continuous signal as

$$X(z)=\sum_{n=0}^{\infty} x_n z^{-n} \qquad (12)$$

A FIR filter generates a new digital signal y_n from x_n using delay, multiple and addition operations as shown in figure-12 and eqn. (13)

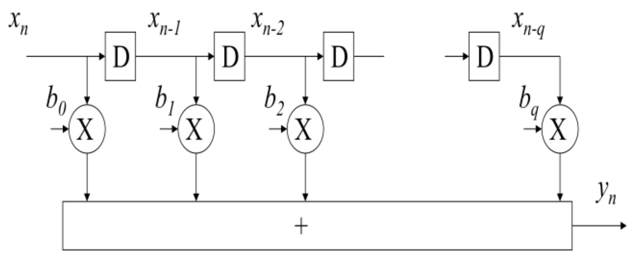

Figure 12: Block Diagram of FIR Filter

$$y_n = x_n b_0 + x_{n-1}b_1 + x_{n-2}b_2 + ...x_{n-q}b_q = \sum_{i=0}^{q} x_{n-i}b_i \qquad (13)$$

where b_i are known as the filter coefficients and delay D is equal to the sample, symbol period T. Taking Z Transform,

$$\begin{aligned} Y(z) &= X(z)b_0 + X(z)z^{-1}b_1 + X(z)z^{-2}b_2 + ...+ X(z)z^{-q}b_q \\ &= X(z)\left(b_0 + z^{-1}b_1 + z^{-2}b_2 + ...+ z^{-q}b_q\right) \\ &= X(z)\sum_{i=0}^{q} z^{-i}b_i \end{aligned} \qquad (14)$$

Where z^{-n} may be taken to mean a delay of n sample periods. Y(z)=X(z)H(z). Hence the transfer function H(z) is

$$H(z)=\frac{Y(z)}{X(z)}=\sum_{i=0}^{q} z^{-i}b_i \qquad (15)$$

A Recursive Infinite Impulse Response Filter(IIR) generates a new digital signal y_n from the input x_n as shown below in the figure-13 and eqn (16).

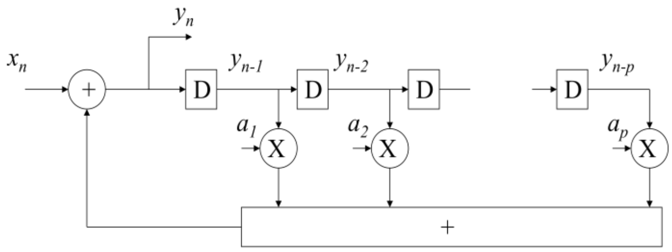

Figure 13: Block Diagram of IIR Filter

$$y_n = x_n + y_{n-1}a_1 + y_{n-2}a_2 + ...y_{n-p}a_p = x_n + \sum_{i=1}^{p} y_{n-i}a_i \qquad (16)$$

Where a_i are known as the filter coefficients and delay D is equal to the sample symbol period T.

Taking Z Transform yields,

$$Y(z)=X(z)+Y(z)z^{-1}a_1+Y(z)z^{-2}a_2+.....+Y(z)z^{-p}a_p$$

Rearranging,

$$X(z)=Y(z)-Y(z)z^{-1}a_1-Y(z)z^{-2}a_2-.....-Y(z)z^{-p}a_p$$

$$\begin{aligned} \mathrm{X(z)} &= Y(z)\left(1-z^{-1}a_1-z^{-2}a_2-.....-z^{-p}a_p\right) \\ &= Y(z)\left(1-\sum_{i=1}^{p} z^{-i}a_i\right) \end{aligned}$$

$$H(z)=\frac{Y(z)}{X(z)}=\frac{1}{\left(1-\sum_{i=1}^{p} z^{-i}a_i\right)} \qquad (17)$$

Let the received signal be p(t) and in digital form, $p_n=p(nT)$.

Zero equaliser means zero ISI. This implies that the Equaliser $H_E(z)$ is non-zero in response to pulse n at sample instant, i.e the filter output is the unit pulse δ_n in response to p_n. Z transform of δ_n is equal to 1. $P(z)H_E(z) = 1$; $H_E(z)= 1/P(z)$.

$$\mathrm{P(z)} = \sum_{i=0}^{\infty} p_i z^{-i}$$

$$H_E(z)=\frac{1}{P(z)}=\frac{1}{p_0z^0+p_1z^{-1}+p_2z^{-2}+.....}=\frac{1}{\sum_{i=0}^{\infty} p_i z^{-i}} \qquad (18)$$

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This has the form of IIR Filter which is difficult to deal with due to instability and cumbersome derivation. The simplest solution is to use an FIR approximation with truncation to ideal impulse response. This will give rise to significant errors. FIR filter equaliser output can be written in time domain with the Zero Forcing constraint as $y_n=1$ for one value of n and $y_n=0$ otherwise. We have $q+1$ filter coefficients. We set up $q+1$ equations and unknowns and solve for the coefficients. Thus equalisation will reduce ISI and hence increase the eye opening. The frequency response of a digital filter is obtained by substituting, $z = e^{jwT}$. At frequencies where $P(e^{jwT})$ is small, large noise amplification will occur. The received pulse spectrum and equalizer spectral response are as follows.

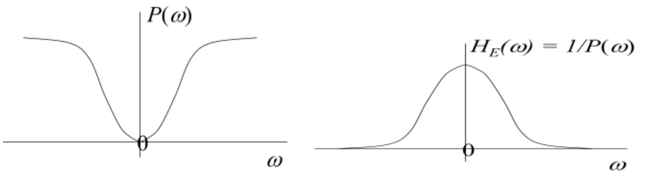

Figure 14: Spectral Response of Receiver and Equaliser

Low Pulse Spectrum response near zero will give rise to high gain and noise enhancement by the equaliser in this region. The eye diagram implemented in with and without

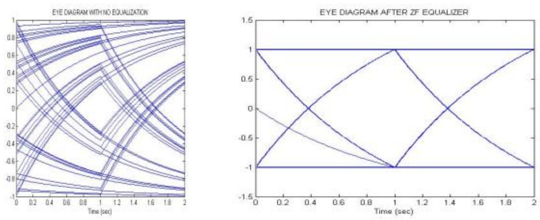

Figure 15: Eye Diagram of ZF Equaliser without Noise.

ZF equalisation, without noise will be as above. The ZF Equaliser in AWGN environment is implemented and compared as follows.

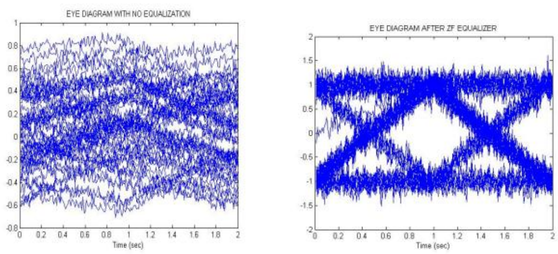

Figure 16: Eye Diagram of ZF Equaliser with Noise.

B. LMS EQUALISER

With Mean Square Error (MSE) criteria between the received signal, h[n] and the desired signal, x[n], equaliser filter response, θ can be designed to give Least Squares(LS) Algorithm and Least Mean Square(LMS) algorithm.

$$\begin{aligned} J[\theta] &= x^T x - x^T H\theta - \theta^T H^T x + \theta^T H^T H\theta \\ &= x^T x - \underbrace{2x^T H\theta}_{scalar} + \theta^T H^T H\theta \end{aligned} \tag{19}$$

$$\frac{\partial J(\theta)}{\partial \theta} = -\underbrace{2H^T x}_{scalar} + 2H^T H\theta \tag{20}$$

$$\hat{\theta} = (H^T H)^{-1} H^T x \tag{21}$$

The LMS system block diagram can be given as below in figure 17:-

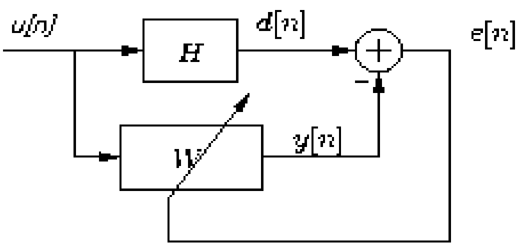

Figure 17: Block Diagram of LMS Equaliser.

Let *u[n]* be the input signal from channel, *d[n]* be the desired response, *h[n]* be the training sequence generator, *W* be the FIR Filter tap weights factor and *e[n]* be the error feedback between desired response and Equaliser output. Steepest Descent algorithm is used in LMS to find recursively the best solution for the cost function. Gradient is a changing vector either to increase or decrease the step-size, for the adaptive filter to converge. The Mean Square Error is equal to

$$E\{[(d(k)-y(k)]^2\} = E\{[(d(k)-\sum_{n=-N}^{N} w(n)u(k-n)]^2\} \tag{22}$$

$$\begin{aligned} &E\{[(d(k)-\sum_{n=-N}^{N} w(n)u(k-n)]^2\} = E\{d(k)^2\} - 2\sum_{n=-N}^{N} w(n)P_{du}(n) + \sum_{n=-N}^{N}\sum_{m=-N}^{N} w(n)w(m)R(n-m) \\ &P_{du}(n) = E\{d(k)u(n-k)\} \\ &R_{uu}(n-m) = E\{u(m-k)u(n-k)\} \end{aligned} \tag{23}$$

In order to obtain LMS Minimum MSE, we can derive and compare it to zero to get the MMSE. Results are as follows.

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$$\frac{d(MSE)}{dW(k)} = \frac{d(E\{d(k)^2\} - 2\sum_{n=-N}^{N} w(n)P_{du}(n) + \sum_{n=-N}^{N}\sum_{m=-N}^{N} w(n)w(m)R(n-m))}{dW(k)}$$

$$w_{opt} = R^{-1} \bullet P \qquad (24)$$

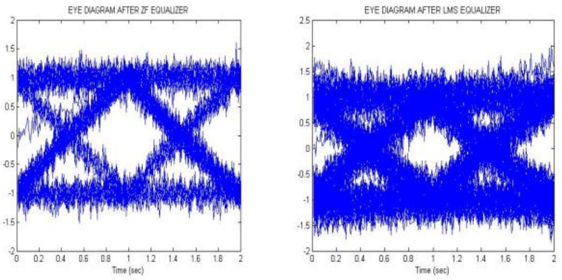

Figure 18: Eye Diagram of ZF Vs LMSEqualiser with Noise

LMS has the advantage of simplicity of implementation, non-negligence of noise and bypassing Inverse matrix. However, it has slow convergence and requirement of training sequence, which will reduce Communication Bandwidth.

C. NON LINEAR - DECISION FEEDBACK EQUALISER

DFE contains a feed-forward(FFF) and feedback filter(FBF) each. There will be 'L' forward weights and (N-L) feedback weights. In each symbol period, equaliser receives K input samples at FFF as well as one decision / Training sample at FBF. It outputs a weighted sum in delay lines and updates the weights to prepare the next symbol. The block diagram is shown below at figure.19.

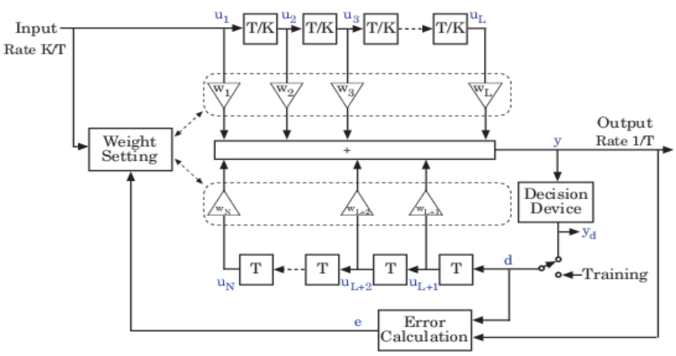

Figure 19: Block Diagram of DFE Equaliser

In MLSE Equaliser, the conditional probabilities are used for making decision on a sequence of symbols. Bit-error-rate with DFE and MLSE are computed and the results are shown below at figure.20.

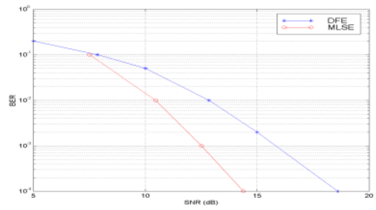

Figure 20: BER of DFE Vs MLSE

FBMC- EQUALISERS

The above equaliser techniques were implemented on FBMC Communications on ITU Vehicular Channel A and Pedestrain B Channel. Results of Bit Error Rate in MMSE equalizers are compared and found that the FBMC-MSE Equalisers have better BER results on a given SNR. The results are given below at figure.21.

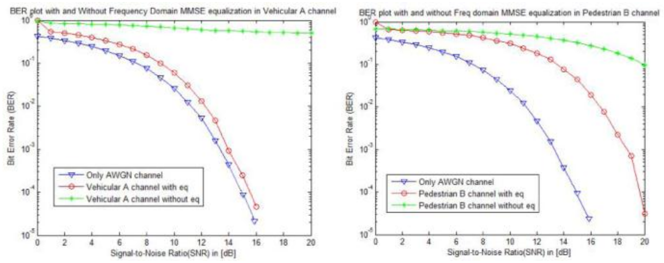

Figure 21: BER results of FBMC-MSE Equaliser with Noise

Similarly, FBMC with MMSE results with Iterative convergence are implemented and compared. For implementation, 16-QAM with N=64 symbols and M=4 Overlapping factor and channel Length of L=16 are considered. The ISI Power is reduced by observing over a window of L and obtained min MMSE. Further by taking average ISI Power over the window, the Avg MMSE was calculated. The comparisons are shown below at figures.22 &23.

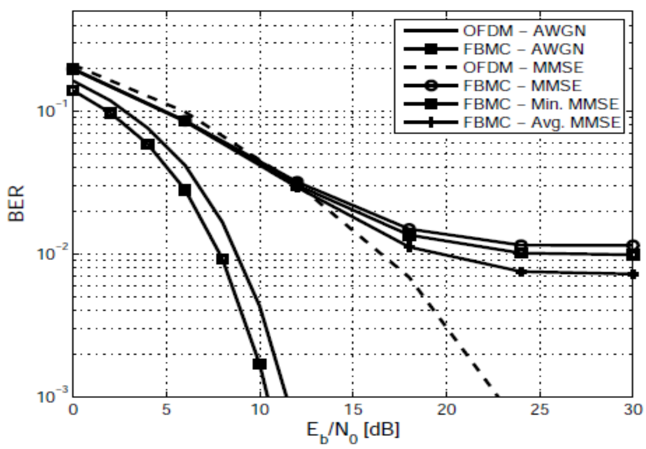

Figure 22: BER Vs E_B/N_o results of FBMC-MMSE Equaliser.

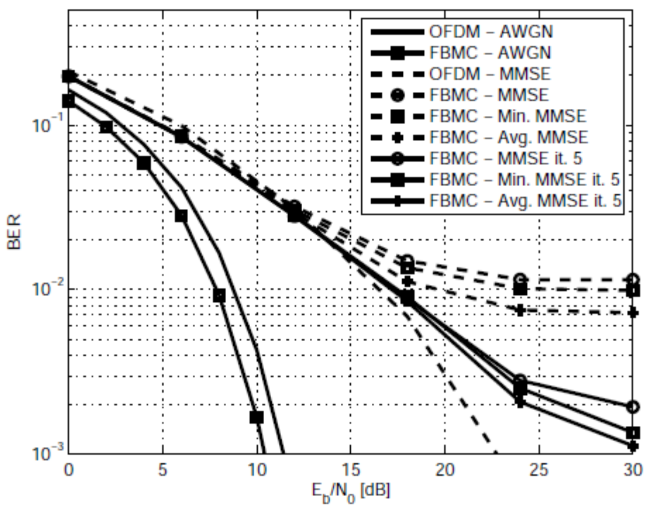

Figure 23: BER Vs E_B/N_o of Iterative FBMC-MMSE Equalisers.

With the above comparisons in Bit Error Rate, Spectrum Efficiency, Computations, FBMC with minimum MMSE Equaliser is the best candidate in comparison with OFDM for 5G Communication.

CONCLUSION AND FUTURE WORKS

The present 4 G Communication Technology based on OFDM is prone to spectrum leakage, strict Orthogonality conditions among entire sub-carriers. Its rectangular pulse-shape increases susceptibility to synchronisation errors. In order to mitigate the major drawbacks, new Communication technology based on FBMC techniques are likely for future SDR. It has more spectrum efficiency, more flexibility to use different pulse shape and efficient power control. This FBMC Systems can also be susceptible to ICI (Interference of Symbols from other subcarriers in the same index) and ISI (Interference of Symbols in different time index from all subcarriers) depends upon Pulse-shape, Channel and Data. It can be mitigated through Channel Equalisation methods suggested in this paper. Among the all, Minimum Mean Square Error Techniques are most promising for inclusion as standards of 5 G. In future Channel Estimation and Synchronisation techniques will find developments and improvements.

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