International Journal of Digital Information and Wireless Communications (IJDIWC) 3(4): 400-410
The Society of Digital Information and Wireless Communications, 2013 (ISSN: 2225-658X)

Adaptive Channel Equalization for FBMC Based on Variable Length Step Size and Mean-Squared Error

Mashhoor AlTarayrah and Qasem Abu Al-Haija
King Faisal University, Department of Electrical Engineering,
Al-Ahsa 31982, P.O. Box 380, Saudi Arabia
Emails: Mtarayrah@kfu.edu.sa; Qalhaija@kfu.edu.sa

ABSTRACT

Recently, increasing data transmission rates and the demand of more bandwidth at the same time have been a challenge. The trend now is to support high data rates in wireless communications. Multicarrier systems have overcome many challenges of high bandwidth efficiency and at the same time provided also high spectral efficiency. Filter bank multicarrier systems (FBMC) provide some advantages more than the traditional orthogonal frequency division multiplexing (OFDM) with cyclic prefix (CP). FBMC systems provide a much better spectral shaping of the subcarriers than orthogonal frequency division multiplexing (OFDM). Therefore, the most obvious difference between the two techniques is in frequency selectivity. In this paper, we will present a least-mean-square (LMS) algorithm which is based on well-known cost functions, which is the mean-squared error (MSE) adapted for FBMC system with offset QAM modulation (OQAM). This leads to a per-subchannel adaptive equalizer solution with low complexity. The proposed simulations have used practical channel information based on the International Telecommunications Union (ITU) Standards. Moreover, we will discuss how the proposed algorithm will optimize and evaluate the convergence characteristic curves of LMS equalization algorithm per-subcarrier.

KEYWORDS

Filter Banks, Multicarrier Modulation, Offset Quadrature Amplitude Modulation (OQAM), Per-Subchannel Equalization, LMS- Algorithm.

1 INTRODUCTION

Wireless communication systems are amongst the most complicated communication systems which arise essentially from the hostile nature of wireless channels that are far from being linear or time invariant. Other sources of complications are the market need for user mobility, global roaming, diverse high quality services, and high data rate multimedia services. These constraints guide naturally to complex transmitter and receiver architectures [1], [2], and [7].

Wireless communications need to support high data rate, with high quality to transmit data, thus requires a wide transmission bandwidth. Increasing the transmission rate generally converts the communication channels into a frequency selective one. Frequency selectivity appears in the form of inter-symbol interference (ISI) that results from the generated multipath effect [1], [5]. The following schematic in figure 1 illustrates how the inter-symbol interference is generated when the signal bandwidth is large (this is the case for high-bit rate signals). The condition for frequency selectivity for the signal bandwidth (BT) is much larger than the channel coherence bandwidth (Δf_c), i.e.

$$B_T \gg \Delta f_c \tag{1}$$

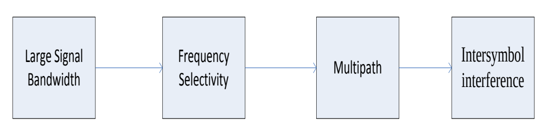

Figure 1: ISI generating in large signal bandwidth

Multicarrier Modulation (MCM) is an efficient transmission technique that consists of splitting up the channel bandwidth at a higher symbol rate into M parallel lower rate sub-channels each with narrower band. MCM divides the available channel bandwidth into M sub-channels through the use of M narrowband subcarriers [7].

Figure 2 illustrates the MCM principle where $s(n)$ represents a bit stream of the input data to be transmitted over the channel which will be divided into blocks of length b to be allocated to each subcarrier such that the k_{th} subcarrier has a set of b_k bits. Thus the total number of bits b per block can be expressed as:

$$b = \sum_{k=0}^{M-1} b_k \qquad (2)$$

The number of bits b_k can vary from one subcarrier to another. To increase the bit rate we can increase the number of bits in subcarriers that have higher signal to noise ratios (SNR) [2].

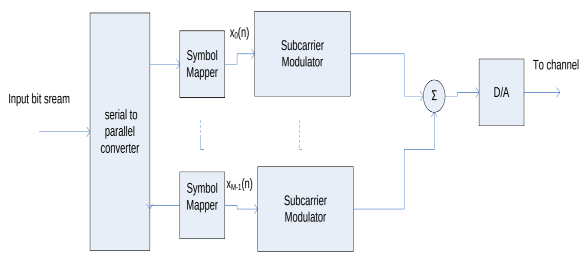

Figure 2: Multicarrier modulation

Multicarrier systems provide many attractive properties for high rate wireless communications. A major advantage of the multicarrier approach is its robustness to the multipath effect, and therefore to ISI where multicarrier modulation splits the large-bit rate incoming sequence into several parallel lower-bit rate sequences. The number of parallel sequences can be adjusted such that the wireless channel becomes frequency non-selective. Filter bank based multicarrier (FBMC) systems offer a number of benefits over conventional multicarrier systems based on Orthogonal Frequency Division Multiplexing (OFDM) [2], [6], [7].

The most obvious difference between the two techniques is frequency selectivity. OFDM exhibits large ripples in the frequency domain. In contrast, FBMC divides the transmission channel of the system into a set of sub-channels, where any channel overlaps only with its immediate neighbors as shown in Figure 3 [2], [6].

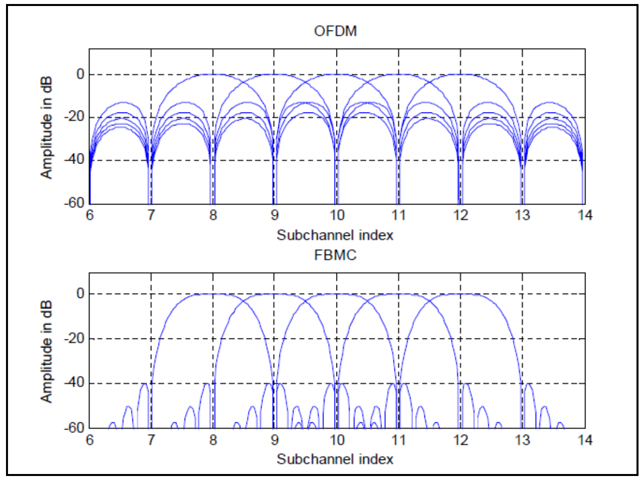

Figure 3: Comparison of sub-channel filter magnitude responses in the case of OFDM and FBMC.

Channel equalization is a technique that can be used to improve received signal quality and link performance as it is used to combat ISI. In equalization process there are two important tasks, the first one is to mitigate the ISI effect, and the other is to prevent enhancement noise power in the received signal in the processing of ISI mitigation. These tasks must be balanced in frequency selective channel equalization. Since the channel is random and time varying, equalizer must deal with time varying characteristic of the channel and so it is called adaptive equalizer or equalizer training. The adaptive algorithm is based on the error signal where derivative of the signal based on comparing between the equalizer outputs $\hat{d}_k$ with referenced signal d_k which comes from the replica of the transmitted signal x_k. The adaptive algorithms use error (e_k) to update the equalizer weights in such a way that reduces the cost function iteratively [4].

Generally, the channel equalization plays a major role in enhancing the performance of communication systems such as FBMC system which uses simple sub-channel equalization, where equalizers take the form of FIR filters [1] with very small numbers of taps. It should be emphasized that using multicarrier modulation leads to a substantial simplification of the required channel equalizers; due to its ability to combat the multipath/ISI effects of the frequency selective wireless channel [5].

2 LITERATURE REVIEW

Tarayrah and Abu Al-Haija in [1] (2013) proposed a preliminary study of this paper as a reconfigurable design of channel equalization algorithms for FBMC system with offset QAM modulation (OQAM). They optimized their algorithms based on the mean-squared error (MSE) criterion and the Least-Mean-Square (LMS) and applied them to each sub-carrier which resulted in an adaptive equalizer with lower complexity.

Yahia Medjahdi et. al. in [8] (2011) provided a theoretical performance evaluation of the downlink of asynchronous orthogonal frequency division multiplexing (OFDM) and filter bank based multicarrier(FBMC) cellular radio communication systems, and they developed an accurate derivation for the interference caused by the timing synchronization errors in the neighboring cells. Their system considered the multipath effects on the interfering and desired signal are also considered as well as the frequency correlation fading in the case of block subcarrier assignment scheme. Finally, they derived the exact expressions for average error rates of OFDM and FBMC systems.

Tero Ihalainen et. al. in [9] (2011) studied the channel equalization in filter bank multicarrier (FBMC) transmission based on the offset quadrature-amplitude modulation (OQAM) subcarrier modulation. They derived the finite impulse response (FIR) per-subchannel equalizers based on the frequency sampling (FS) approach, both for the single-input multiple-output (SIMO) receive diversity and the multiple-input multiple-output (MIMO) spatially multiplexed FBMC/OQAM systems. Their FS design consisted of computing the equalizer in the frequency domain at a number of frequency points within a subchannel bandwidth and deriving the coefficients of subcarrier-wise equalizers. Also, they evaluated the error rate performance and computational complexity for both antenna configurations and compared them with the SIMO/MIMO OFDM equalizers. The results obtained confirmed the effectiveness of their proposed technique with channels that exhibit significant frequency selectivity at the subchannel level and showed a performance comparable with the optimum minimum mean-square-error equalizer, despite a significantly lower computational complexity.

Tero Ihalainen et. al. in [10] (2010) presented a new approach to generate an FBMC signal waveform focusing on the contiguous narrowband subcarrier allocation, which is often encountered in uplink transmission. Their proposed scheme relies on a cascade of aP-subchannel synthesis bank ($P << M$), time domain interpolation, and a user-specific frequency shift. Their approach provided a notable computational complexity savings over a wide range of practical user bandwidth allocations, when compared to the conventional implementation consisting of equally sized filter banks. Also, their novel scheme provided a flexible and low-complexity synthesis of spectrally well-localized FBMC uplink waveforms with a strong potential for future broadband mobile communications.

Tobias Hidalgo Stitz et. al. in [7] (2010) presented a detailed analysis of synchronization methods based on scattered pilots for filter bank based multicarrier(FBMC) communications, taking into account the interplay of the synchronization, channel estimation, and equalization methods.

They showed that by applying pilots designed specifically for filter banks, the carrier frequency offset (CFO), fractional time delay (FTD), and channel response can be accurately estimated. Also, they performed the channel parameter estimation and compensation are in the frequency domain, in a subchannel-wise fashion. Finally, the performance evaluation was applied in a hypothetical WiMAX scenario in which an FBMC system would substitute OFDM maintaining as much physical layer compatibility as possible.

Eleftherios Kofidis and Athanasios A. Rontogiannis in [11] (2010) proposed an adaptive T/2-spaced decision-feedback equalization (DFE) algorithm for MIMO-FBMC/OQAM systems, that is both computationally efficient and numerically stable. The structure of their algorithm followed the V-BLAST idea where the algorithm was applied in a per subcarrier fashion. Their simulation results reported that the proposed algorithm demonstrated its effectiveness in time-varying MIMO channels with high frequency selectivity.

Aıssa Ikhlef and Jerome Louveaux in [12] (2009) investigated the problem of equalization for filter bank multicarrier modulation based on offset OQAM (FBMC/OQAM). They found that the existing equalizers for FBMC/OQAM do not completely cancel inter carrier interference (ICI) and hence some ICI remains even after equalization. To cope more efficiently with ICI, they proposed a two stages MMSE (TS-MMSE)equalizer; the first stage consisted in applying an MMSE equalizer and taking some preliminary decisions, then subsequently using these decisions to remove the term corresponding to ICI from the received signal for each subchannel. The resulting signal from the first stage has its ICI practically removed (up to channel estimation errors and decision errors) and is therefore corrupted only by inter-symbol interference (ISI) and additive noise. In the second stage they applied an MMSE equalizer which copes only with ISI. The proposed two stage MMSE equalizer had shown better performance compared to the classical MMSE one.

Leonardo G. Baltar, Dirk S. Waldhauser and Josef A. Nossek in [13] (2009) presented a per-subchannel nonlinear equalizer for a class of filter bank based multicarrier (FBMC) systems. They considered the class of exponentially modulated FBMCs with offset quadrature amplitude modulated (OQAM) input symbols and then designed a fractionally spaced decision feedback equalizer (DFE) that minimizes the mean squared error (MMSE) and taking into account the inter-(sub)channel interference (ICI). Their simulation results showed that despite its increased computational complexity, the performance and the higher bandwidth efficiency of OQAM FBMC systems makes them a competitive alternative to conventional multicarrier systems like cyclic prefix based orthogonal frequency division multiplexing (CP-OFDM).

Y. Medjahdi et. al. in [14] (2009) focused on the downlink of multi cellular networks and investigated the influence of the inter-cell interference in an unsynchronized frequency division duplex (FDD) context with a frequency reuse of 1. They compared the conventional orthogonal frequency division multiplexing with cyclic prefix modulation (CP-OFDM) and the filter bank based multi-carrier modulation (FBMC). Finally, they evaluated the performance in terms of average capacity in FBMC multi-cell networks compared to CP-OFDM ones.

Ari Viholainen et. al. in [15] (2009) discussed an efficient prototype filter design in the context of filter bank based multicarrier (FBMC) transmission. They analyzed the performance of various designs using the offset-QAM based FBMC system and provided numerical results to characterize different optimization criteria in terms of frequency selectivity of resulting prototype filters and total interference level of the filter bank structure. Finally, they have shown the kind of performance trade-offs that can be obtained by adjusting those free parameters which offered a useful information to a system designer.

Aıssa Ikhlef and Jerome Louveaux in [16] (2009) studied the distortions of multiple-input multiple-output (MIMO) filter bank multicarrier modulation (FBMC) systems such as inter-symbol interference (ISI), and inter-carrier interference (ICI), and using multiple antennas creates inter antenna interference (IAI), which are caused by the frequency selective transmission channel. To mitigate these distortions, they derived an MMSE equalizer assuming spatial multiplexing is used and they proposed a successive interference cancellation (SIC) and Ordered SIC (OSIC) techniques to extract the transmitted streams as well as they improved the performance by introducing a two stage OSIC (TSOSIC) technique. Their simulation results confirmed the effectiveness of the proposed techniques over the classical one tap equalizer.

Dirk S. Waldhauser, Leonardo G. Baltar and Josef A. Nossek in [17] (2008) presented a least-mean-square (LMS) algorithm adapted to the principle of orthogonally multiplexed QAM filter banks(OQAM-FBMC) which led to an adaptive equalizer solution with low complexity. The initialization of the LMS equalizer resulted from a pilot based channel estimation. They compared their results with a classical OFDM system, where the loss in data rate is compensated with a higher modulation scheme.

In this paper, we propose to design an adaptive channel equalization algorithm for FBMC system with offset QAM modulation based on the well-known cost functions Mean-Squared Error (MSE) criterion. We will design the LMS equalizer to every subcarrier in order to gain the benefits of using different step size values for each subcarrier.

3 PROPOSED SYSTEM MODEL

A figure 4 contains the general block diagram of FBMC system which will be used in this paper and the main processing blocks are: OQAM pre-processing, Synthesis filter bank (SFB), Analysis filter bank (AFB), OQAM post–processing and LMS per-subcarrier equalization. We assumed Linear Time Varying (LTV) channel model which is based on AWGN channel and has physical characteristic like multipath propagation. We focused on a specific prototype filter length that is ($L_p = KM$), an extra delay z^{-D} has to be introduced on the SFB or AFB input, here depends on L_p, that is $L_p = KM + 1 - D$, so in our model the extra delay is z^{-1}[1].

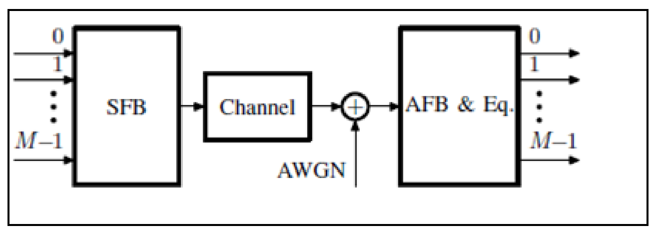

Figure 4: FBMC system

3.1 OQAM PREPROCESSING

The first block is OQAM preprocessing block which converts the QAM symbols into OQAM. There are two steps to convert QAM symbols into OQAM, firstly, a simple complex to real conversion required. We must know that the conversion will be different for even and odd sub-channels as shown in Table 1. This conversion increases the sample rate by 2 [1], [5], and [7].

Secondly, the conversion is followed by multiplication $\theta_{m,n}$ sequence, where

$$\theta_{m,n} = j^{m+n} \quad (3)$$

n is discrete time variable that runs at twice the rate of l. The pattern of real and imaginary samples must follow the sign of the $\theta_{m,n}$ sequences. Here $\theta_{m,n}$ sequences can be as follows as an example [7]:

$$\theta_{m,n} = \begin{cases} 1, j, 1, j, 1, .. & \text{for n is even} \\ j, 1, j, 1, j, .. & \text{for n is odd} \end{cases}$$

After the OQAM preprocessing, the input signals are either pure real or pure imaginary.

Converting the QAM symbols to OQAM format involves two important specificities [5]:

- A time offset of half a QAM symbol period $(T/2)$ is applied to either the real part or the imaginary part of the QAM symbol when the OQAM signal is generated.
- For two successive sub-channels, say m and $m+1$, the offset are applied to the real part of the QAM symbol in sub-channel m, while it is applied to the imaginary part of the QAM symbol in sub-channel $m+1$.

The OQAM symbols will be denoted by $\beta_{m,n}$, OQAM symbols are given by:

$$\beta_{m,n} = \theta_{m,n} d_{m,n} \quad (4)$$

Table 1: OQAM symbol $\beta_{m,n}$ for even/odd values of m&n

	Even n	Odd n
Even m	$c^{R}_{m,n/2}$	$c^{I}_{m,(n-1)/2}$
Odd m	$c^{I}_{m,n/2}$	$c^{R}_{m,(n-1)/2}$

3.2 SYNTHESIS FILTER BANK

Analysis filter bank consists of serial to parallel converter (delay chain and down-samplers by M/2), poly-phase filtering, FFTs, and simple multipliers as shown in figure 5.

$$\beta_{k,n} = (-1)^{kn}(-1)^{kK} e^{\left(j\frac{\pi k}{M}\right)} \quad (5)$$

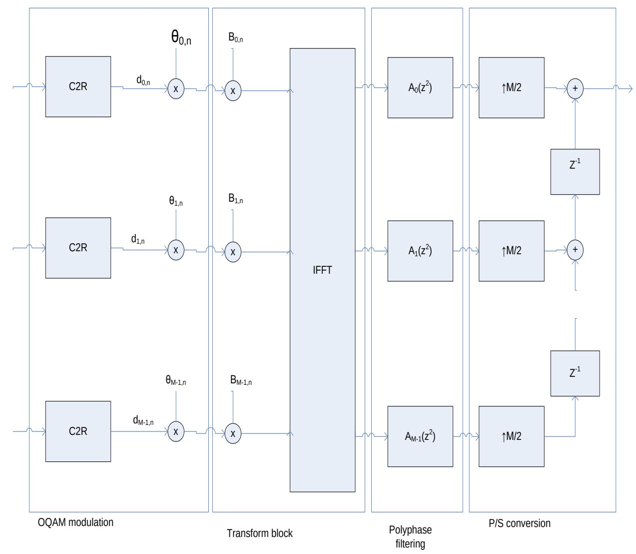

Figure 5: Poly-phase filter bank structures Synthesis filter bank (SFB).

The output signal of SFB $s[m]$ is complex-valued. We can express the discrete-time baseband signal at the output of the SFB of an FBMC transmitter based on OQAM modulation as

$$s(m) = \sum_{k=0}^{M-1} \sum_{n=-\infty}^{\infty} d_{k,n}\theta_{k,n} g_k(m - nM/2) \quad (6)$$

Where

$$\theta_{k,n} = j^{(k+n)}$$

$$d_{k,n} = (-1)^{kn}$$

And M is number of subcarriers, $d_{k,n}$ is the real-valued symbol at the k^{th} subcarrier during the n^{th} symbol interval, $g_k(m)$ are shift-invariant impulse responses of the SFB channel filters.

3.3 ANALYSIS FILTER BANK

Synthesis filter bank that consists of simple multipliers, IFFT, poly-phase filtering, and parallel to serial converter (up-samplers by M/2 and delay chain) as shown in figure 6.

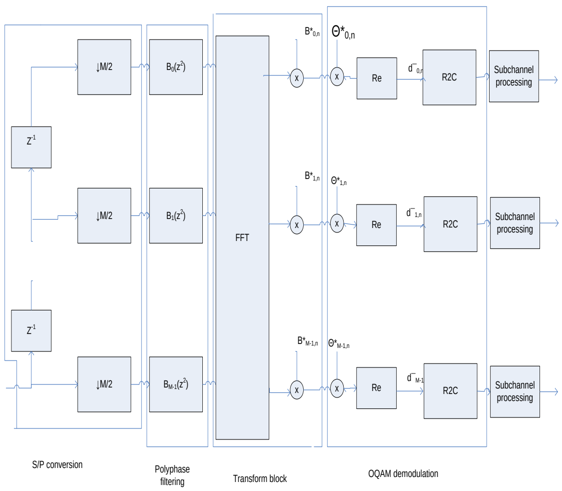

Figure 6: Poly-phase filter bank structures Analysis filter bank (AFB).

Then AFB output is

$$\hat{\beta}_{k,n} = (-1)^{kn} e^{\left(-j\frac{2\pi k}{m}\left(\frac{L+1}{2}\right)\right)} \qquad (7)$$

And because the length of the prototype filters is $L = KM$. Then

$$\hat{\beta}_{k,n} = (-1)^{kn}(-1)^{kK} e^{\left(-j\frac{\pi k}{M}\right)}$$

$$\hat{\beta}_{k,n} = \beta^{*}_{k,n} \qquad (8)$$

Here the extra delay is merged to S/P converter.

3.4 OQAM POST-PROCESSING

In the post-processing operation there are 2 steps, firstly, the real part should be taken after multiplication by $\theta^{*}_{m,n}$ sequence. The second operation is real-to-complex conversion, where two successive real-valued symbols (with one multiplied by j) form a complex-valued symbol $\hat{c}_{m,l}$. This conversion decreases the sample rate by a factor 2 [3, 7].

3.5 LMS PER-SUBCARRIER EQUALIZATION

Sub-channel equalization in this paper is based in MSE criterion, and the LMS algorithm is used as adaptive equalizer, here a per-subcarrier equalizer works at $T/2$, where $T$is the symbol duration. One FIR equalizer per subcarrier is used to mitigate ISI and ICI that results from the frequency selective channel and to improve the symbol decisions. The LMS equalizer is used in the minimization of the mean square error (MSE) between the desired output of the equalizer and the actual output equalizer [1], [7].

In this paper we applied the equalizer to the real and imaginary parts for each sub-carrier individually. The filter weights are updated as in the below equation. LMS can be calculated iteratively by

$$\hat{d}_k(n) = w_N^T(n) y_N(n) \qquad (10)$$

$$e_k(n) = x_k(n) - \hat{d}_k(n) \qquad (11)$$

$$w_N(n+1) = w_N(n) - \mu_u * e_k^*(n) y_N(n) \qquad (12)$$

Now to compute the mean square error $|e_k|^2$ at instance time k

$$MSE = E[e_k^* e_k] \qquad (13)$$

3.6 THE COMMUNICATION CHANNEL

Wireless communication channels can affect the input signals in different mechanisms, such as linear and nonlinear distortion, additive random noise, fading and interference. The communication channel can be modeled as a linear time invariant system with transfer function $C(z)$ followed, by a zero-mean additive white Gaussian noise (AWGN) source$e(n)$. But we can't apply this model directly to wireless communications channels, because it is time-varying. In fact, we can represent a short-term time-invariant for a wireless communication to be useful in most cases of practical interest. Figure 7 shows a simplified block diagram of a communication channel [6].

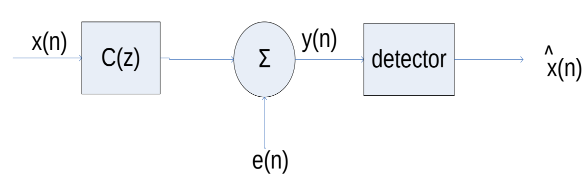

Figure 7: Linear time-invariant communication channel model

$x(n)$ Assumed to be consisted of the sum of M sequences, each transmitted in one of M sub-channels. Mathematically, we'll have

$$x(n) = \sum_{i=0}^{M-1} x_i(n) \qquad (14)$$

The output of the channel can be as follows:

$$y(n) = x(n) * c(n) + e(n) \qquad (15)$$

The received signal $y(n)$ is a noisy and distorted signal version of the input signal $x(n)$. $\hat{x}(n)$ is the estimated value which is the output of the detector, but may it will be not identical to the transmitted signal because of ISI caused by channel which is because of multipath fading and limitation in bandwidth, and noise $e(n)$, this will give us a nonzero probability of error P_e.

In wireless communication, the channels are time varying so a single $C(z)$ can't be used to represent them successfully [6].

3.7 DOWNSAMPLERS/UPSAMPLERS

Up-samplers are part of the SFB which increases the sampling rate, the $\frac{M}{2}$- fold up-sampler inserts $\frac{M}{2} - 1$ zeroes between adjacent samples of its input signal. The up-sampling operation is illustrated as in Figure 8 [6].

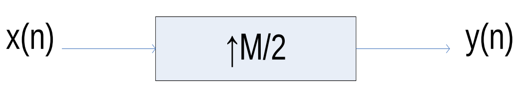

Figure 8: M/2 -fold up-sampler

The input-output relationship of the up-sampler can be expressed as

$$y(n) = x(2n/M) \qquad (16)$$

Down-sampler is part of the AFB, which reduces the sampling rate. The $M/2$ down-sampling operation is illustrated as in Figure 9 [6].

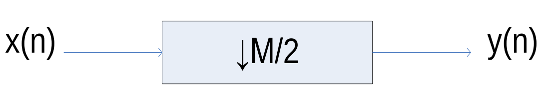

Figure 9: $M/2$-fold down-sampler

An -fold down-sampler performs the input-output relation

$$y(n) = x(\frac{nM}{2}) \qquad (17)$$

4 SIMULATION ENVIRONMENTS

We have used a computer simulation software package-MATLAB to simulate the performance of the OQAM FBMC system applying the per-subcarrier LMS equalizer. We have assumed 1MHz channel bandwidth with $M = 1024$ sub-channels and the overlapping factor $K = 4$ based on OQAM modulation. We have assumed Linear Time Varying (LTV) channel model which is based on AWGN channel and has physical characteristic like multipath propagation and we have used in our simulation practical channel information based on the International Telecommunications Union ITU Standards (Vehicular channel type A) and 12 dB SNR. In the LMS sub-channel equalizer we have used 2 different step size equalizer for several sub-channels and 3 tap equalizer.

5 RESULTS AND COMPARISONS

The following simulation results illustrate the performance of sub-channel equalizers which were evaluated by the computer simulation software package-MATLAB as shown in the flowchart of figure 12. Prototype filter designed with the sampling frequency technique $(L = K * M)$ coefficients.

In design the prototype filter we have used simple technique which is called frequency sampling technique, and it is presented with the following parameters

$$L = 4096, M = 1024, K = 4.$$

We started the design by determination of L desired values $H(k/L); 0 \leq k \leq L-1$ in the frequency domain by

$$H(0) = 1$$
$$H(1/L) = 0.97195983;$$

$$H(2/L) = \sqrt{1}/2;$$
$$H(3/L) = 0.23514695;$$
$$H\left(\frac{k}{L}\right) = 0. \; for\; 4 \leq k \leq L-1 \qquad (18)$$

Then, the prototype filter coefficients are obtained by inverse DFT as

$$h(m) = 1 + 2\sum_{k=1}^{K-1}(-1)^k H(k/L)\cos(2\pi km/L). \quad for\; 1 \leq m \leq L-1$$

Where

$$h(0) = 0$$

In fact, the condition $h_1 = 0$ determines the desired values $H(1/L)$ and $H(3/L)$. It is necessary to make the number of coefficients an odd number, so that the filter delay is an integer number of sample periods. The Impulse response of the prototype filter is shown in figures 10 and 11 show the frequency response obtained. In this figures, the sub-channel spacing Δf is taken as unity (Δf =1).

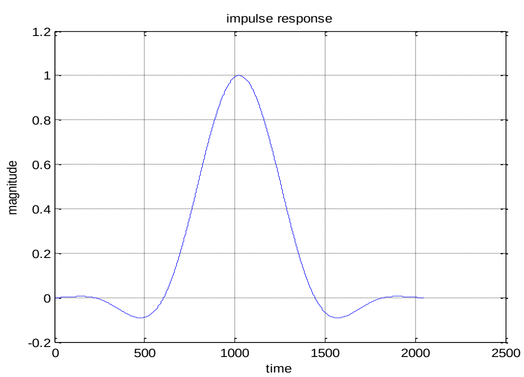

Figure 10: Impulse Response of the prototype filter

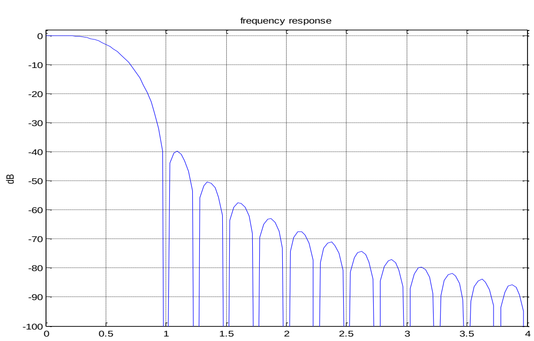

Figure 11: Frequency response of the prototype filter

It is important to notice that the filter attenuation exceeds 60 dB for the frequency range above 2 sub-channel spacings.

Another important feature is the background noise of the system, due to the non-orthogonality of the sub-channels with even indices with respect to the sub-channel with zero index. With this prototype filter, the background noise is $\sigma_b^2 = -65\,dB$.

Finally, the convergence characteristics of LMS algorithm were performed. In the simulation results N represents the equalizer taps and Mu represents the step size of the equalizer.

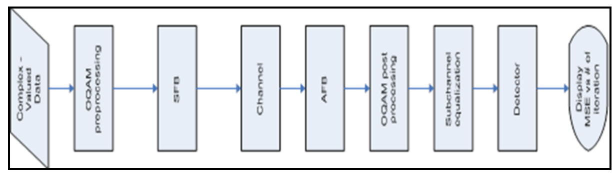

Figure 12: Flowchart of Sub-channel equalization based on FBMC (Read Left – To- Right)

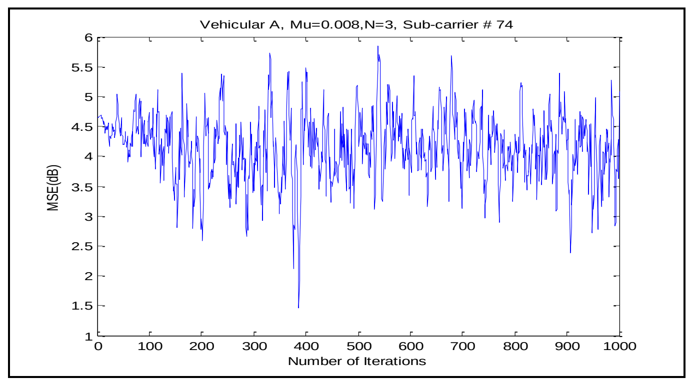

Figure 13: MSE vs. Number of Iteration for Vehicular a channel

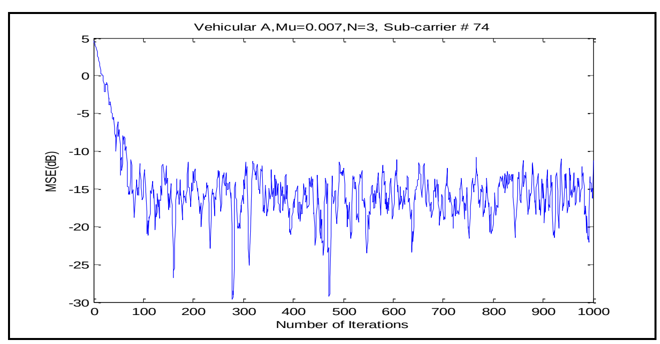

Figure 14: MSE vs. Number of Iteration for Vehicular a channel, with 0.007 step size

For subcarrier number 74 the convergence characteristic curves of LMS algorithm shown in

Figures 13 and 14. It was taken to Vehicular channel type A, 3 tap equalizer and 12 dB SNR. We noticed that the MSE in Vehicular A channel model with step size 0.007 converges to 18 dB which is better than with 0.008 which converges to 4 dB.

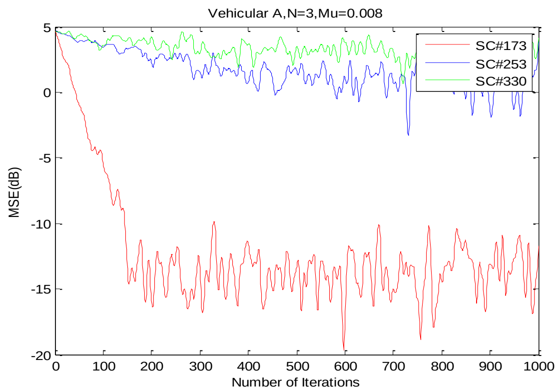

Figure 15: Comparison of MSE vs. Number of iteration for different subcarriers,Mu=0.008

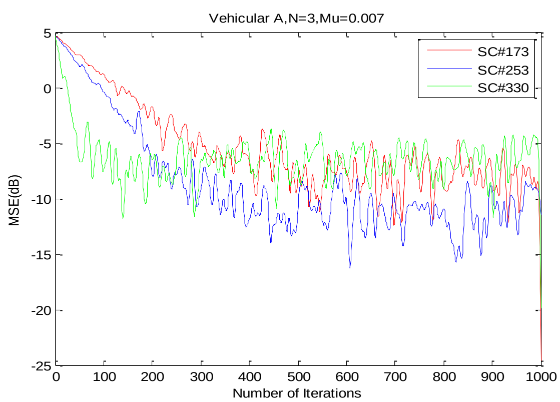

Figure 16: Comparison of MSE vs. Number of iteration for different subcarriers ,Mu=0.007

Comparsion between the convergence characteristic curves of LMS algorithm for different subcarriers (173,263 and 330) and different step size of the equalizers is shown in Figures 15 and 16

We noticed that it converges quickly for subcarrier 173 with 0.008 step size to -15 dB and for step size $M_u = 0.007$ it converges to -10 dB. Howeever, for subcarrier 330 with 0.007 step size it converges quickly to -10 dB and with $M_u = 0.008$ step size it converges to 4 dB.

We noticed from previous results that the same subcarrier could have a good characteristic curve at sertain step sizes of the equalizer and a bad characteristic curve at another step sizes . At the same time another subcarrier will be in opposite way with the same step size valuse in the first subcarrier. And that is because of frequency selectivity which give each subcarrier different correlation matrix which should the step size to be taken.

6 CONCLUSIONS

Using FBMC systems eleminate Intercarrier Interference ICI and ISI and convert the channel to frequency selective one therefore no need for using complex equalizer thus simple adaptive LMS equalizer is enough. And because of Using FBMC systems the data rates have been increased for large bandwidth as this a demand in wilreless communication.

We have designed adaptive channel equalization algorithms for FBMC systems with offset QAM modulation and perfect channel estimation, We have optimized the mean squared error (MSE) criterion by using simple adaptive LMS equalizer because of its simplicity , as LMS per-subchannel equalizer operates as fractionally spaced (T/2) equalizer which aims to avoid irrevocable aliasing of the subchannels. Also, we have optimized and compared the convergence curve of LMS equalization algorithms as well as applied the equalizer to practical channel information based on ITU Standards (VEH A) and 12 dB SNR with different equalizer step sizes and subcarriers. Because of frequency selectivity, every sub-channel will have a different optimum LMS equalizer step sizes based on the correlation matrix to every sub-channel as illustrated in the simulation results.

ACKNOWLEDGEMENT

Authors appreciate the publication support of Deanship of Scientific Research at King Faisal University KFU-AL-AHSA.

7 REFERENCES

1. Mashhoor Al Tarayrah and Qasem Abu Al-Haija, "LMS-based Equalization in Filter Bank Multicarrier Wireless Communication Systems", The International Conference on Digital Information Processing, E-Business and Cloud Computing (DIPECC2013), UAE, pp. 87-9,Octobar 2013.
2. Tero Ihalainen, Yuan Yang and Renfors M., “Filter Bank Based Frequency-Domain Equalizers with Diversity Combining” IEEE International Symposium on Circuits and Systems, pp. 93-96, 2008.
3. John Proakis and Masoud Salehi, Digital Communications, Fifth Edition, McGraw-Hill, 2008.
4. P.Siohan, C.Siclet, and N.Lacaille, “Analysis and Design of OFDM/OQAM Systems Based on Filter bank Theory” IEEE transactions on signal processing, VOL. 50, NO. 5, pp. 1170-1183, MAY 2002.
5. Tero Ihalainen ET. al., “Channel equalization in filter bank based multicarrier modulation for wireless communications” EURASIP Journal on Advances in Signal Process., 2007.
6. P. P. Vaidyanathan, “Filter Banks in Digital Communications” IEEE Transactions on Circuits and Systems 2001; 1(2): 4-25.
7. Tobias Hidalgo Stitz, Tero Ihalainen, A. Viholainen, and M. Renfors, “Pilot-Based Synchronization and Equalization in Filter Bank Multicarrier Communications,” EURASIP Journal on Advances in Signal Processing, 2010; 18 pages.
8. Yahia Medjahdi, et. Al.,"Performance Analysis in the Downlink of Asynchronous OFDM/FBMC Based Multi-Cellular Networks", IEEE transactions on wireless communications, vol. 10, no. 8, august 2011.
9. Tero Ihalainen, et.al. , "Channel Equalization for Multi-Antenna FBMC/OQAM Receivers", IEEE transactions on vehicular technology, vol. 60, no. 5, June 2011.
10. Tero Ihalainen, et.al. , "Generation of Filter Bank-Based Multicarrier Waveform Using Partial Synthesis and Time Domain Interpolation", IEEE transactions on circuits and systems: regular papers, vol. 57, no. 7, July 2010.
11. Eleftherios Kofidis and Athanasios A. Rontogiannis, " Adaptive BLAST Decision-Feedback Equalizer FOR MIMO-FBMC/OQAM Systems", IEEE 21st International Symposium on Personal Indoor and Mobile Radio Communications (PIMRC), 2010.
12. Aissa Ikhlef and Jerome Louveaux, "An Enhanced Mmse Per Subchannel Equalizer For Highly Frequency Selective Channels For FBMC/OQAM Systems", IEEE 10th Workshop On Signal Processing Advances In Wireless Communications, 2009. SPAWC '09.
13. Leonardo G. Baltar, Dirk S. Waldhauser and Josef A. Nossek, "MMSE Subchannel Decision Feedback Equalization for Filter Bank Based Multicarrier Systems ", IEEE Circuits and Systems, 2009.
14. Yahia Medjahdi, et. Al., "INTER-CELL INTERFERENCE ANALYSIS FOR OFDM/FBMC SYSTEMS ", on Signal.
15. Ari Viholainen et. al., "PROTOTYPE FILTER DESIGN FOR FILTER BANK BASED MULTICARRIER TRANSMISSION", 17th European Signal Processing Conference (EUSIPCO 2009).
16. Aissa Ikhlef and Jerome Louveaux, " Per subchannel equalization for MIMO FBMC/OQAM system", on Communications
17. Dirk S. Waldhauser, Leonardo G. Baltar and Josef A. Nossek, "Adaptive Equalization for Filter Bank Based Multicarrier Systems" IEEE Circuits and Systems, 2008.