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Frequency Domain Adaptive Equalization of Multipath Fast Fading Channel in OFDM System

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Abstract— Adaptive equalization in frequency domain mitigates the effect of inter-symbol interference in OFDM system which caused by non-line of sight in wireless communication environment, where the channel is characterized as a multipath channel. Adaptive channel estimation based on superimposed training sequence is an efficient technique that provides fast tracking capability, without loss in bandwidth efficiency, in situations where the channel is time varying. Such technique avoids the use of frequency multiplex training pilots with the transmitted data symbols. In this paper, we investigate the adaptive channel estimation method in frequency domain for time varying Rayleigh fading channel using Superimposed Training Sequence (STS pilot) and Recursive Least Square (RLS) adaptive algorithm.

Keywords— OFDM, Equalization, Superimposed training sequence, RLS.

1 Introduction

Orthogonal frequency division multiplexing (OFDM) system have attracted much attention as an efficient technology in wireless communication systems [1]. In OFDM systems whole spectrum is divided into several sub-carriers, and before each OFDM block the cyclic prefix is inserted. So OFDM systems can mitigate the effect of multipath and have high spectrum efficiency [2].

Many channel estimation methods in OFDM systems are proposed up to now. Blind channel estimation schemes offer a high spectral efficiency, but they converge slower and required more complex computation [3] as like in time domain channel estimation [4]. Pilot symbols aided modulation (PSAM) [5] which inserting known symbols in time domain or frequency domain can provide channel estimation but degrade the bandwidth utilization. Thus channel estimation based on superimposed training sequence were proposed [6] [7]. The main feature of superimposed pilot technique is to save the bandwidth and give an effective tracking for multipath fast fading channels.

The presence of Fast Fourier Transform (FFT) in OFDM system make it perfect to apply the frequency domain adaptive equalization in which the adaptive filters in each frequency bins can be applied also it has many advantages as compared with time domain equalization, perform multiplication rather than convolution, length of adaptive filter being decimated, computational complexity being reduced , block size can be smaller, provide increase in convergence speed (this is result from decreased eigen value spread of autocorrelation matrix of the signals in the filter update), also in frequency domain the idea of adaptive filter can be applied with a low power DSPs. All these features were attracted to perform the adaptive channel estimation and equalization in frequency domain in spite of the increment in memory requirements.

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Adaptive channel estimation algorithms are being widely deployed in channel estimation to track channel under wireless random time varying conditions. The adaptive algorithms after successive iterations converges to the optimum solution and these requires received signal of pilot information, which were sent at transmitter [8]-[10]. LMS algorithm requires less computational and it is easy to implement while RLS algorithm requires more computational and give best convergence in practical. The convergence analysis of the algorithm plays a major role in application of equalization in OFDM under fast fading channels. We always seeking to find a method way to improve the convergence and applicable to fast fading channels. Recursive least squares (RLS) adaptive filter utilized for the purpose of OFDM channel equalization.

2 System Model

The generalized wireless transceiver using OFDM, as shown in figure 1, modulates the input binary sequence onto PSK/QAM mapper and then performs IFFT on parallel converted data. In our work BPSK, QPSK, 16 QAM and 64 QAM were considered as per IEEE 802.11a standards. Discrete-time model baseband equivalent OFDM system with $\boldsymbol{N}$ subcarriers experiencing selective multipath fading channel. Let $\boldsymbol{N_{cp}}$ denote the CP length which is inserted to avoid intersymbol interference and $\boldsymbol{X} = [\boldsymbol{S}(0) + \boldsymbol{P}(0)\ \boldsymbol{S}(1) + \boldsymbol{P}(1) \ldots \boldsymbol{S}(N-1) + \boldsymbol{P}(\mathrm{N}-1)]^{\mathrm{T}}$ indicate the summation of modulated complex symbols represented by $\boldsymbol{S}$ with superimposed training sequence represented by $\boldsymbol{P}$ for an OFDM symbol. By N-point inverse fast Fourier transform (IFFT), the transmitted complex baseband signal is:

$$x(n) = \frac{1}{\sqrt{N}} \sum_{k=0}^{N-1} X(k) e^{j2\pi k \frac{n}{N}} \tag{1}$$

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Where $n = -N_{cp}, -N_{cp}+1, \ldots, N-1$. Let the channel impulse response $h_l(n)$ be the time varying channel gain of the lth path at time instant n, where $l = 0, 1, \ldots, L-1$, in which the maximum time-delay spread (L) is less than or equal the CP length (N_{cp}). The synchronizations of timing and frequency are assumed to be perfect. At the receiver side, after removing CP, the received signal is obtained as:

$$y(n) = \sum_{l=0}^{L-1} h_l(n)x(n-l) + z(n) \tag{2}$$

Where n = 0, 1, …, $N-1$, and z(n) represents the complex additive white Gaussian noise (AWGN) with zero-mean and variance σ_Z^2.

At the receiver, after removing the cyclic prefix (CP), the signal model for each block can be expressed as

$$\boldsymbol{y} = \boldsymbol{h}\mathbf{x} + \boldsymbol{z} \tag{3}$$

Where $\mathbf{x}$ is the $(N \times 1)$ transmitted block, $\boldsymbol{h}$ is an $(N \times N)$ circulant matrix with first column $[h_0, h_1, \ldots, h_{L-1}, 0, \ldots, 0]^T$ which represent channel matrix in time domain
$\boldsymbol{z}$ is $(N \times 1)$ the Gaussian noise vector; $(.)^{\mathrm{T}}$ represents the transpose operator.
After FFT the received signal in the frequency domain is

$$\mathbf{Y} = \mathbf{HX} + \mathbf{Z} \tag{4}$$

Where $\boldsymbol{Y}, \boldsymbol{X}$ and $\boldsymbol{Z}$ are the frequency domain notation for the received signal data symbols and noise respectively with all dimensions are $(N \times 1)$, $\mathbf{H}$ channel matrix with dimensions $(N \times N)$. Applying a proper compensation technique can make the diagonal values of this matrix represents the channel frequency response without intercarrier interference (ICI) response. Under such circumstances the off-diagonal values of this matrix are made zeroes and a one tap equalizer becomes suitably possible to be applied for channel estimation, $\mathbf{H} = \mathrm{diag}[\boldsymbol{H}(0)\ \ \boldsymbol{H}(1) \ldots \boldsymbol{H}(N-1)]$. Otherwise, a further advanced equalization technique is required [11].

3 Superimposed Training Sequence (STS)

Many works explained that the estimation of the channel at the certain pilot frequencies which is pilot aided channel estimation in OFDM which is excellent way for channel estimation in slow fading channel but till now the main challenge is how to treat the fast fading environment which is the main focus of our work.

A technique were used in adding pilots in the transmitter known as Superimposed Training Sequence (STS) where pilots are added to the data symbols in this way saves valuable bandwidth at the expense of a reduction in the information signal-to-noise ratio (SNR), since some of the transmitted energy is allocated to the hidden pilots. STS schemes offer tradeoffs between loss of rate (slots for training) and simplicity of the receiver, channel estimation vs. tracking, and possibly improved power efficiency.

Going back to equation (1) we will find that $X(k) = data + STS$ so let S denote the notation of data and P denote the periodic training sequence so the transmitted power ratio following the below relationship:

$$X = \sqrt{1-\beta}\, S + \sqrt{\beta}\, P \tag{5}$$

Where β is the power ratio between data and superimposed training sequence $0 \le \beta \le 1$, and related to the value of pilot amplitude α.

$$P = \sum_{m=0}^{R-1} \alpha\delta(n - mK) \tag{6}$$

Where $\delta(.)$, α And K are the Kronecker delta function, the Pilot Amplitude and the pilot period respectively.

The channel coefficients can be consistently estimated using the first-order statistics of the received signal. In order to simplify channel estimation, P is often chosen to be periodic. A disadvantage of this method is that the performance of the channel estimator is affected by the embedded unknown data, which acts like input noise to the adaptive channel estimation. In order to better explain this effect and also to motivate the proposed STS scheme, we use the following frequency domain interpretation the fast Fourier transform (FFT) of $\boldsymbol{y}$ can be written

$$\mathbf{Y} = \mathbf{HS} + \mathbf{HP} + \boldsymbol{Z} \tag{7}$$

Where $\mathbf{H} = \mathrm{diag}[\boldsymbol{H}(0)\ \ \boldsymbol{H}(1) \ldots \boldsymbol{H}(N-1)]$ with $H(n)$ being the frequency response of the channel at normalized frequency $2\pi k/N$. The energy of the data symbols is spread over all frequency bins. The channel coefficients are then estimated using the pilot frequencies, treating **HS** and **Z** as additive noise sequences. Here, we propose to develop a channel estimator that is completely impervious to the unknown data.

4 Adaptive Channel Estimation and Equalization

The main problem in wireless communication systems is signal fading. The channel estimation techniques and equalization methods for tracking the original signals are most important for wireless communication system.

The transmitted signal will be distorted and suffer from intersymbol interference (ISI) due to multipath fading channel, therefore the transmitted data cannot recovered in the receiver accurately. When the channel is time varying, channel estimation process is highly desired at the receiver.

We can classify the channel estimation into two broad categories.

i. Inverse channel estimation, as shown in figure 2.
ii. Direct channel estimation, as shown in figure 3.

Inverse channel estimation (Equalization) could be applied in case of single carrier modulations, the OFDM system is preferable where simple one tap frequency domain equalizer (FDE) used to equalize the OFDM signal which pass over frequency selective fading channel. When the channel impulse response remains constant over one symbol period, at each subcarrier the received signal.

The transmitted signal is restored by one tap equalizer as follows:

$$\hat{X}(n) = W(n)Y(n) \tag{8}$$

Where $W(n)$ represents the equalizer coefficients. Without prior information about the channel, the adaptive algorithms can adjust the coefficients of the equalizer to reduce $E\{|\hat{X}(n) - X(n)|^2\}$. The error $e(n)$ signal is obtained by comparing the equalized signal $\hat{X}(n)$ with the reference signal. After that, the coefficients of the equalizer are adjusted in accordance with the error signal

$$W(n+1) = W(n) - g(n)e(n) \tag{9}$$

Where $g(n)$ represents the gain factor, depends on the (RLS) algorithm. In the training mode, the reference signal is known pilot data to the transmitter and the receiver. In which we can restore the distorted signal by adaptive one tap equalizer.

$$\hat{X}(n) = H(n)^{-1}Y(n) \tag{10}$$

5 Simulation Results

Following IEEE 802.11a specification. The transmitted signal will suffer from fast and frequency selective fading due to the Doppler shift and the multipath respectively. The performance is measured through the Bit Error Rate (BER) in the system versus the energy per bit to noise power spectral density (PSD) ratio (Eb/No) which is significant parameter in digital communication system and measured in (dB). Figure 4 show (BER) versus (Eb/No) with different modulation techniques such as (BPSK, QPSK, 16QAM, 64QAM) for the channel estimation, figure 5 represents the learning curve of the used RLS algorithm while figure 6 and figure 7 represent the channel frequency response and channel impulse response respectively.

6 Conclusion

In our work, we present an adaptive channel estimation and equalization method in frequency domain based on superimposed training sequence periodic pilot (STS) using recursive least square estimation (RLS) adaptive algorithm in OFDM system to recognize the time varying Rayleigh fading channel with various modulation schemes such as BPSK, QPSK, 16QAM and 64QAM. This method is simple, less computational, reduce complexity and has high spectral efficiency, the system give good performance even with high Doppler shift and also it minimize the (ISI) effect. The performance results of the investigated method is illustrated by the numerical simulations.

Table 1 Simulation parameters in OFDM system

Variable	Value	Description
FFT	**64**	Fast Fourier Transform size
BW	**20 MHz**	Channel Bandwidth
f_c	**900 MHz**	Carrier Frequency
CP	**16 samples**	Cyclic prefix
NBlocks	**40**	Number of blocks
p	**10**	Pilot Ratio
Chan	**channel**	Rayleigh Fading Channel
L	**40**	Channel Length
λ	**0.9**	Forgetting factor
β	**0.5**	Power ratio between pilot and modulated signal
α	**0.1**	Pilot amplitude
fm	**80 Hz**	Maximum Doppler Shift
D	**1e^{-6}[0 1.75 3.5]**	Multipath Delay vector / three paths
G	**[0 -3 -6] dB**	Multipath Gain vector / three paths

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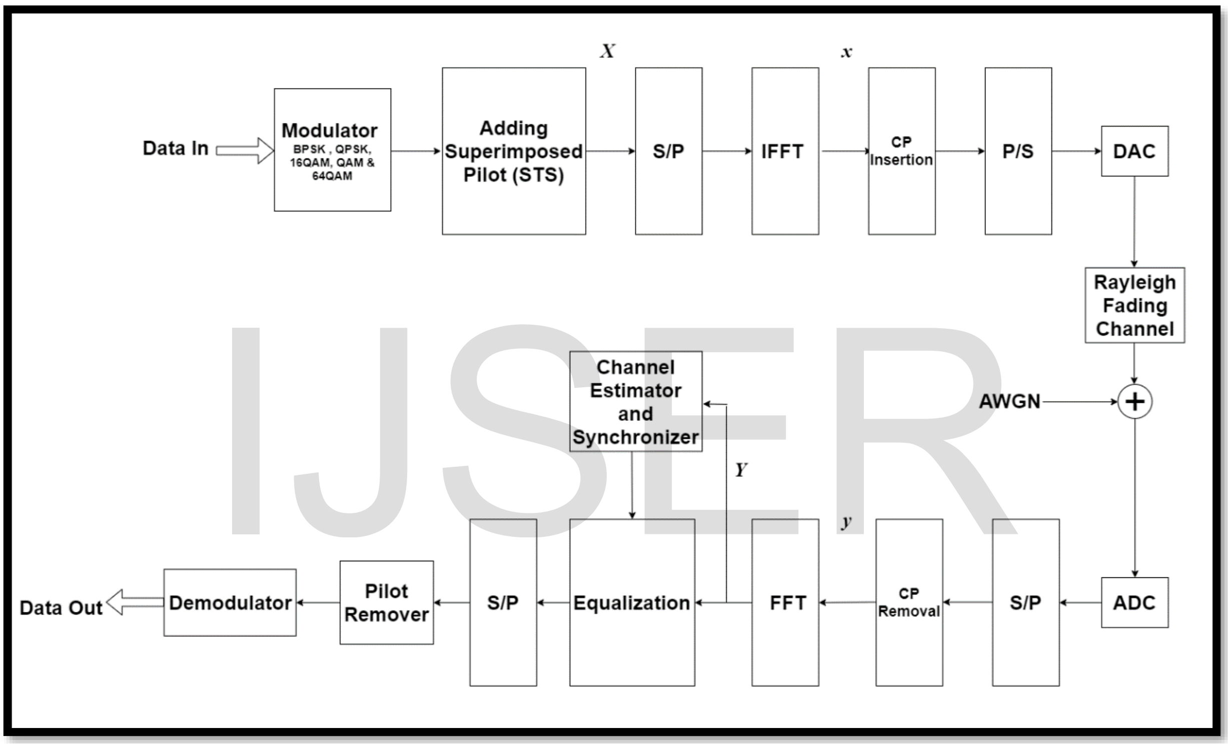

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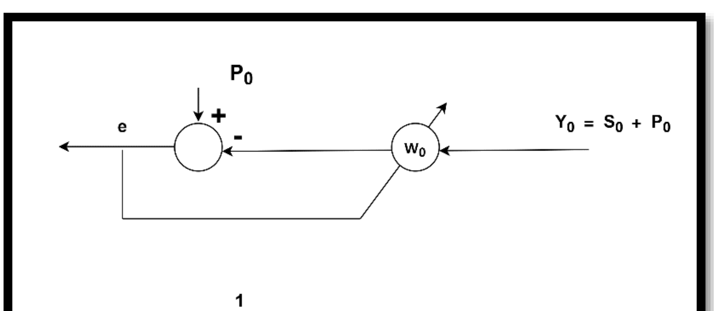

Figure 2: Inverse channel estimation

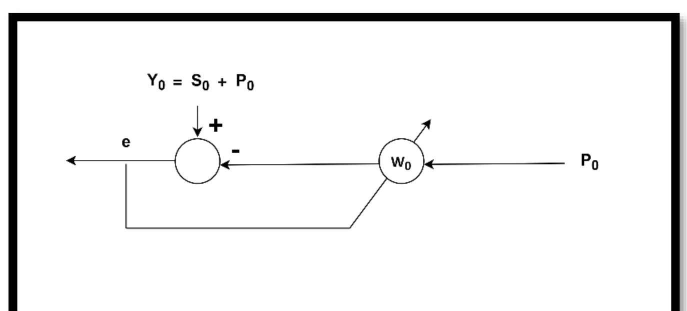

Figure 3: Direct channel estimation

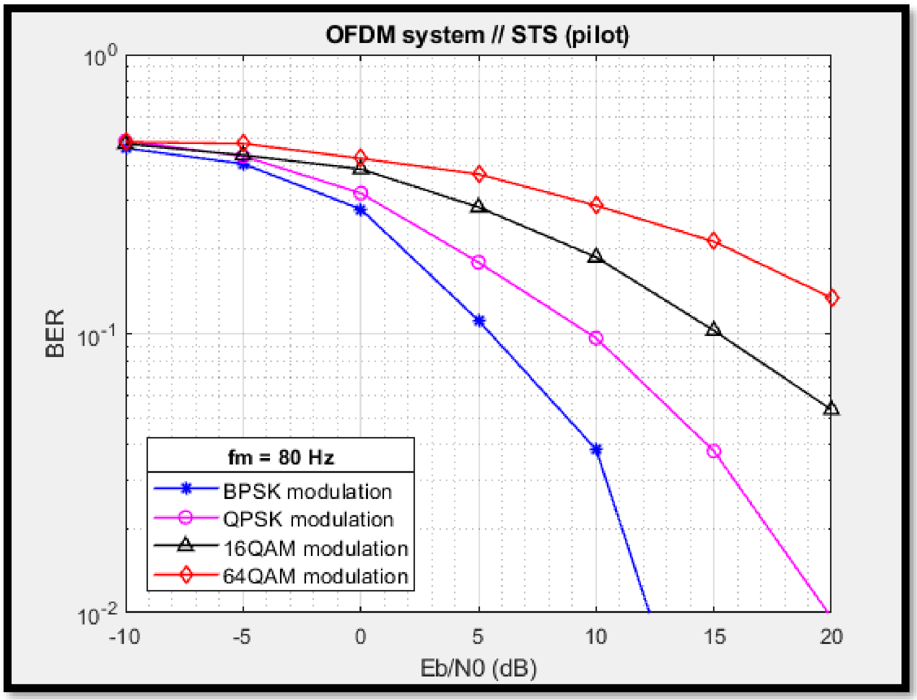

Figure 4: BER versus Eb/N0 with different modulation technique

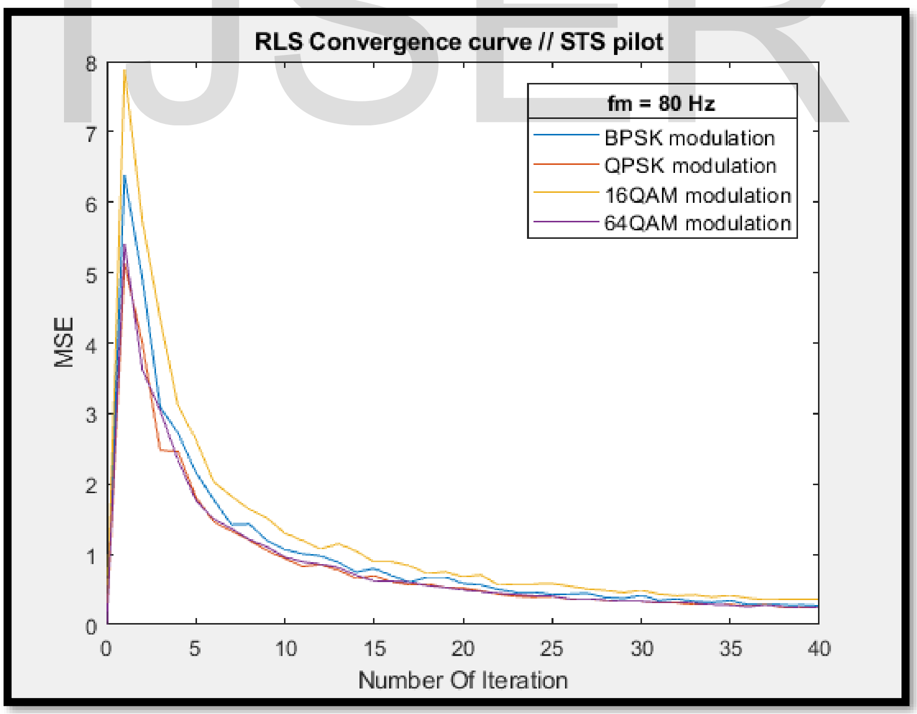

Figure 5: Learning curve of the (RLS) algorithm

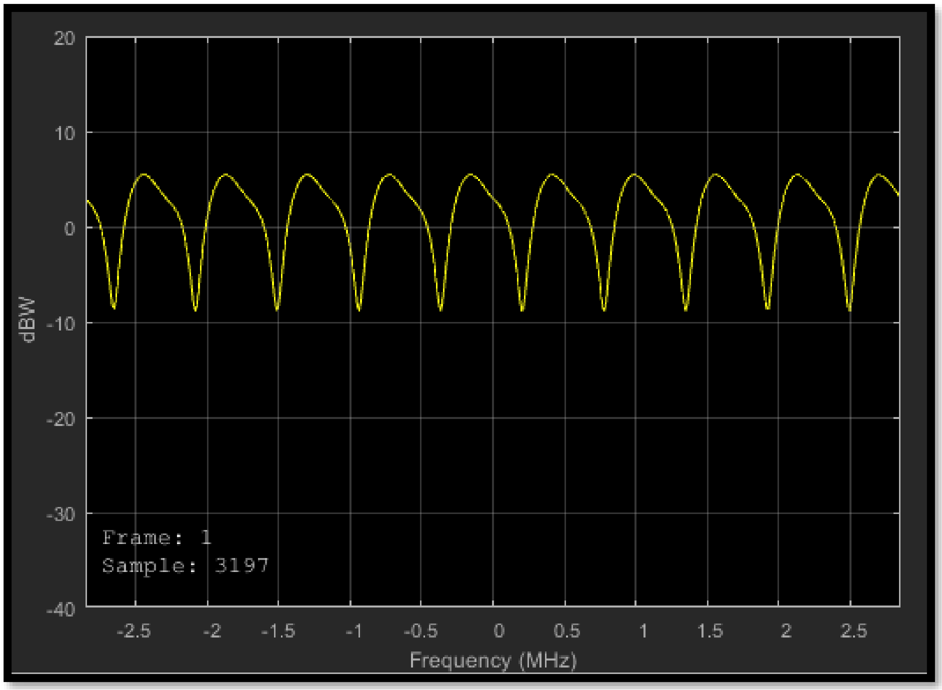

Figure 6: Channel frequency response

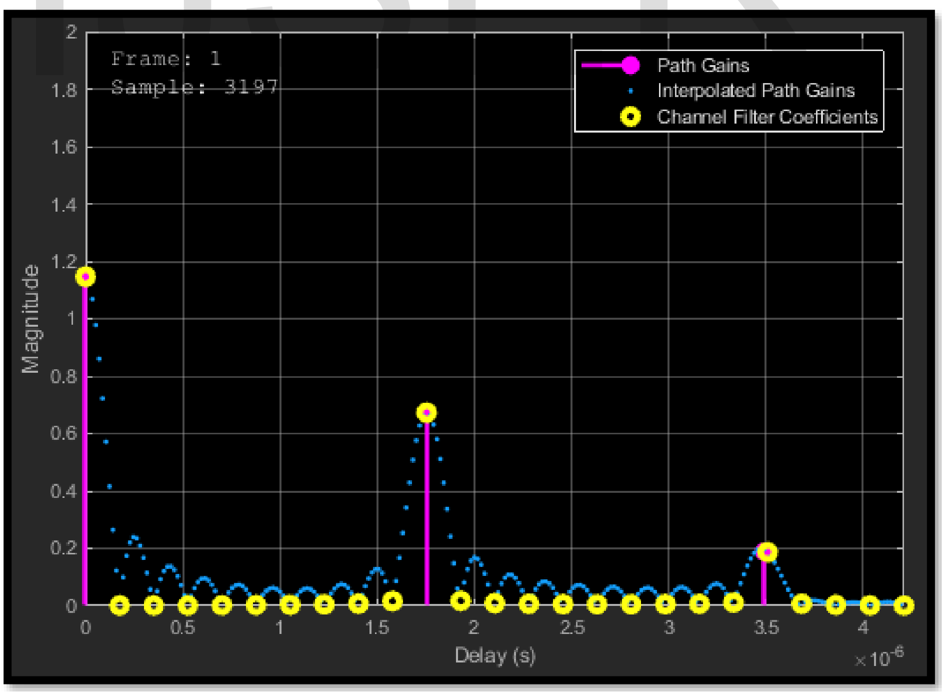

Figure 7: Channel impulse response