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# OFDM Communication Receiver system with Initialisation Processor

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***Abstract*—the task of processing between the two terminals of QAM-Modulator and QAM-Demodulator of the (Orthogonal Frequency-Division Multiplexing) OFDM system implementation, needs special arrangement. If the required data are to be free from (Inter-symbol Interference) ISI and as well as the mitigation of fading, in communication channel. The proposed work, however, is to analyse new type of technique, which can be used to suppress high level noisy signal, acquired from fixed objects (zero-forcing) scenarios existing in the time-variant wireless communication scenarios. This approach, however, will use a technique so called initialisation techniques, in order to eliminate the additive white Gaussian noise (AWGN) in the fading channel, near dc region. While working in the transient mode of operation, with limited number of samples under processing, if digital filter to be used in the receiver side, of the OFDM system. Therefore, the integration of this initialisation technique with the OFDM system was applied to improve the cancellation of the ISI, and further improvement of the signal-to-noise ratio (S/N), by applying the usable-bandwidth technique, in order to decrease the error bit rate of the system.**

***Keywords*- OFDM, QAM-Demodulator, FFT, Initialisation, AWGN, S/N, ISI, Bit-Error-Rate, Fading, Cyclic-Prefix, Parallel-to-Serial.**

## I. INTRODUCTION

**W**ireless systems use multicarrier techniques for high data-rate and to compensate for ISI, in rapidly varying channel. The data stream in these techniques will be divided to, and transmitted in orthogonal sub channels at different subcarriers frequencies. However, if the symbol time is to be chosen as such, it is greater than the delay spread, then ISI will be mitigated. In these systems the modulation and demodulation can be performed by using the Inverse Fast Fourier Transformation (IFFT), and Fast Fourier Transformation (FFT) precise operations as inverse type of process in the transmitter side and receiver side. However, the channel performance is changing from symbol-to-symbol, due to fast fluctuating, to eliminate the error-probability. The OFDM system can employ, the In-Phase and Quadrature, ***I*** and ***Q*** channels. Therefore, a new technique of zero-initialisation and optimum filtering, the narrow band – noise like signal (clutter), to be eliminated. However, a *limiter* level must be rather low with the value of the incoming signal, and linear cancellation is precluded for higher signal faded level, where maximum suppression is required. After converting the signal to a digital words using the A/D converter, two anti-noise or noise-whitening filters (Optimum Bandwidth filters) are utilised to linearly suppress noise, which is latent with the communication channel. A coherent limiter is then used to normalise the noise and the residual ISI, which is operated digitally in ***I*** and ***Q*** channels. An ***initialisation processor*** has been used to enhance the signal-noise like (clutter) rejection capability of the processor when a limited number of samples are considered.

## *II.* OFDM Communication Receiver system with *initialisation Processor*

The principle of operation for each of the blocks will be described as follows;

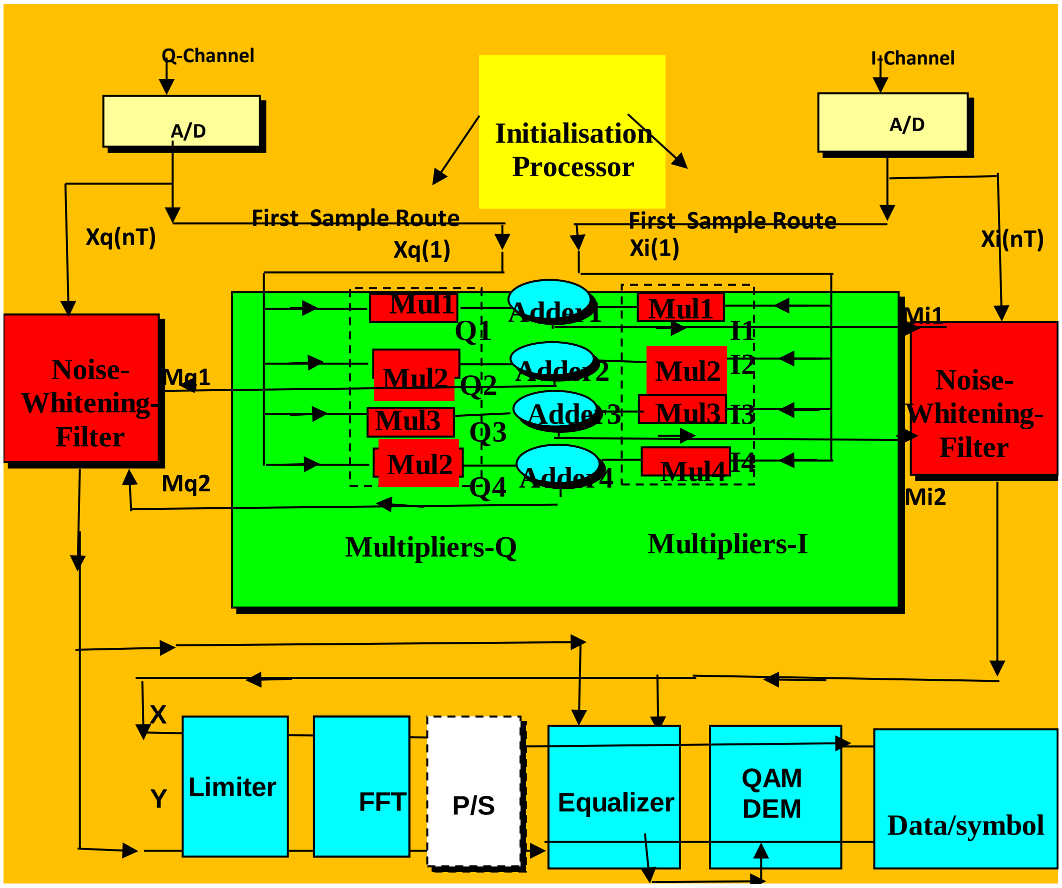


Figure .1 **The** OFDM Communication Receiver system with ***initialisation Processor*** *A*/**D converter**:

The first part of this process is a conversion of the input waveform into ***I*** and ***Q*** signals. The purpose of this conversion is to translate the signal to the baseband. A/D converters generally work on the principle of comparing the voltage of each sample with a succession of progressively higher voltages of precisely known values. When the closest of these is found, the converter produces a binary number equal to the known voltage value.

**B. Initialisation processor**: Initialisation is based on achieving a zero frequency response at **dc** by loading the internal memories of the noise-whitening filters with their steady state values. The configuration of the initialisation processor is as shown in Figure.1. The steady state values used in the initialisation processor are calculated from the first received sample for the ***I*** and ***Q*** channels. These values are loaded to the internal memories of the IIR filters. Therefore, the steady state frequency response at **dc** will be achieved irrespective of the number of processed samples. If the filter has at least one steady state zero at z=1 then the narrow band noisy signal will be attenuated and the desirable signal will be detected. Hence, the recursive IIR filters used can be improved by using the initialisation technique. The improvement is characterised by forcing the response to be zero at zero frequency irrespective of the processed samples. Therefore, this technique is called ***zero-initialisation technique***. A steady state zero will be placed at particular frequency, to induce a variable notch chasing the non-zero Doppler spread noisy signal. The filter considered is a second order filter with a pair of *poles* and *zeros* and its transfer function is given by;

$$H(z) = \frac{a_0 + a_1 z^{-1} + a_2 z^{-2}}{1 - b_1 z^{-1} - b_2 z^{-2}}$$

(1)

Referring to Equation .1, $a_o, a_1, a_2$ are the feed forward coefficients of the filter and $b_1, b_2$ are the feedback coefficients, the internal memories variables $M1(z)$ and $M2(z)$ can be given respectively as;

$$M2(z) = \frac{z^{-1} \times U(z)}{1 - b_1 z^{-1} - b_2 z^{-2}} \quad , \quad \text{and}$$

$$M1(z) = \frac{z^{-2} \times U(z)}{1 - b_1 z^{-1} - b_2 z^{-2}}$$

(2)

The steady state values of the internal memories $M1(z)$ and $M2(z)$ will be evaluated for ***I*** and ***Q*** channels. This evaluation in turn will initialise the nominated filter so that the output will be explicitly zero when the input signal frequency equals to a specific value of $f_z$ . The steady state values for the ***I*** channel can be given as;

$$MI1 = \mathrm{Re}\,al\left[\frac{z^{-2} \times U(z)}{1 - b_1 z^{-1} - b_2 z^{-2}}\right]$$

$$MI2 = \mathrm{Re}\,al\left[\frac{z^{-1} \times U(z)}{1 - b_1 z^{-1} - b_2 z^{-2}}\right]$$

(3)

And the steady state values for the ***Q***-channel can be calculated as;

$$MQ1 = \mathrm{Im}\,aginary\left[\frac{z^{-2} \times U(z)}{1 - b_1 z^{-1} - b_2 z^{-2}}\right]$$

$$MQ2 = \mathrm{Im}\,aginary\left[\frac{z^{-1} \times U(z)}{1 - b_1 z^{-1} - b_2 z^{-2}}\right]$$

(4)

In Equations .2, 3, 4, $U(z) = UI(1) + jUQ(1)$ , where $UI(1)$ and $UQ(1)$ are the first input samples. The independent parameter $z$ is calculated as; $z = \exp(j\omega)$ where $\omega = 2\pi f_z$ .

Therefore, if the arbitrary non-zero frequency initialisation fz , is equal to fd, the corresponding value of a1, and the operator z, can be given as;

a1= -2cos (2*pi*fd)

(5)

Where, fd : is the Doppler frequency to be eliminated , and equals to fz. The value of the operator z, can be given by;

Z= exp (j2*pi*fz*T)

(6)

However, the noisy-signal process C(t) can be expressed by its quadrature components as;

$$\mathrm{C(t)} = \mathrm{C_I(t)}\cos(\omega \mathrm{t}) - \mathrm{C_Q(t)}\sin(\omega \mathrm{t})$$

(7)

Where $\omega$ is containing the noisy-signal carrier frequency

Let $\mathrm{C_I(l)}$ and $\mathrm{C_Q(l)}$ denote the lth Inphase and Quadrature clutter samples in a sequence for $\mathrm{l} = 0,1,\ldots\ldots\ldots\ldots \mathrm{N_I} - 1$ , and consecutively the values of $\mathrm{C_I(l)}$ and $\mathrm{C_Q(l)}$ , can be expressed as;

$$\mathrm{C_I(l)} = \mathrm{m\,C_I(lTs)},$$
$$\mathrm{C_Q(l)} = \mathrm{-m\,C_Q(lTs)}$$

(8)

Where l, is the signal sample taken at t=l Ts, and m, is constant.

Referring to fig1.0, if the noisy-signal is described in the quadrature processor as $\mathrm{C_I(l)}$ and $\mathrm{C_Q(l)}$, the output of filter can be evaluated as given in equation (9), below, by considering the filter transfer function which will be given in equation (9), as for the I, channel;

$$\mathrm{y_I(l)} = \mathrm{C_I(l)} - 2\mathrm{C_I(l-1)} + \mathrm{C_I(l-2)} + \mathrm{b_1 y_I(l-1)} + \mathrm{b_2 y_I(l-2)}$$

(9)

And for the Q-channel, the output of filter can be evaluated as;

$$\mathrm{y_Q(l)} = \mathrm{C_Q(l)} - 2\mathrm{C_Q(l-1)} + \mathrm{C_Q(l-2)} + \mathrm{b_1 y_Q(l-1)} + \mathrm{b_2 y_Q(l-2)}$$

(10)

Equations, 9, and 10, are considering the filter transfer function given by equation 11, used in both I, and Q channels as;

$$H(z)=\frac{1\ -a_1\,z^{-1}+\ \ z^{-2}}{1-b_1\,z^{-1}-b_2\,z^{-2}}$$

(11)

Where , a0=1, a1= -2cos(2*pi*fd) , and a2=1, are the feedforward coefficients of the filter and b1, b2, are the feedback coefficients.

However, if the initialisation technique is considered, the signal to be analysed by initialisation processor can be described as;

$$\mathrm{C_{I,Q}}=\mathrm{C_I}(1)+\mathrm{jC_Q}(l)$$

(12)

Where : $\mathrm{C_I}(1)$ and $\mathrm{C_Q}(l)$ are the first two samples to initialise the internal memories of the designed optimum filter, representing the I-inphase and Q- have two internal memories representing the I-channel as given by;

$$MI1(1)=\mathrm{Re}\,al\left[\frac{z^{-1}\times \mathrm{C_{I,Q}}}{1-b_1z^{-1}-b_2z^{-2}}\right] \qquad \text{And}$$

$$MI\,2(1)=\mathrm{Re}\,al\left[\frac{z^{-2}\times \mathrm{C_{I,Q}}}{1-b_1z^{-1}-b_2z^{-2}}\right] \qquad (13)$$

and the steady state values for the ***Q***-channel can be calculated respectively as;

$$MQ1(1)=\mathrm{Im}\,aginary\left[\frac{z^{-1}\times \mathrm{C_{I,Q}}}{1-b_1z^{-1}-b_2z^{-2}}\right]$$

And

$$MQ2(1)=\mathrm{Im}\,aginary\left[\frac{z^{-2}\times \mathrm{C_{I,Q}}}{1-b_1z^{-1}-b_2z^{-2}}\right] \qquad (14)$$

However, it is worth noting that these steady state values describing I and Q channels representing the second order recursive filter with a transfer function given by equation 11.

It is now convenient to consider the FFT processor for noisy signal only. The kth output coefficient NI(k) for the NI-point FFT in response to the input sequence CI(l) is given by ;

$$\mathrm{N_I}(\mathrm{k})=\sum_{\mathrm{l=0}}^{\mathrm{N_I-1}}\mathrm{C_I}(\mathrm{l})\exp\left(\frac{-\mathrm{j}2\pi\mathrm{kl}}{\mathrm{N_I}}\right) \qquad (15)$$

Where , $\mathrm{k}=0,1,\ldots\ldots\ldots N_I-1$

CI (l) is zero mean Gaussian signal, therefore $\mathrm{E}[\mathrm{C_I}(\mathrm{l})]=0$. Identical results apply for the quadrature component NQ (k). Therefore, if the signal output of the summer follows the FFT is $\mathrm{N}(\mathrm{k})=\mathrm{N_I}(\mathrm{k})+\mathrm{jN_Q}(\mathrm{k})$, then the related real and imaginary parts of N(k) can be expressed as;

$$\mathrm{N}(\mathrm{k})=\mathrm{Re}(\mathrm{N}(\mathrm{k}))+\mathrm{jIm}(\mathrm{N}(\mathrm{k})) \qquad (16)$$

$$\mathrm{N}(\mathrm{k})=\sum_{\mathrm{l=0}}^{\mathrm{N_I-1}}\mathrm{C_I}(\mathrm{l})\cos\left(\frac{2\pi\mathrm{kl}}{\mathrm{N_I}}\right)+\mathrm{C_Q}(\mathrm{l})\sin\left(\frac{2\pi\mathrm{kl}}{\mathrm{N_I}}\right)-\mathrm{j}\sum_{\mathrm{l=0}}^{\mathrm{N_I-1}}\mathrm{C_I}(\mathrm{l})\sin\left(\frac{2\pi\mathrm{kl}}{\mathrm{N_I}}\right)+\mathrm{C_Q}(\mathrm{l})\cos\left(\frac{2\pi\mathrm{kl}}{\mathrm{N_I}}\right) \qquad (17)$$

Where , $\mathrm{k}=0,1,\ldots\ldots\ldots N_I-1$

Equation 17, is describing a zero mean complex Gaussian random variable. Since CI (l) and CQ (l), are zero mean Gaussian random variables. However, defining the envelop r, such that is related to Re{N(k) and Im{N(k)} , and after some manipulations, the density function P(r) is obtained by;

$$\mathrm{P}(\mathrm{r})=\mathrm{r}\exp\left(\frac{-\mathrm{r}^2}{2}\right) \qquad (18)$$

Therefore, probability density function for r, is Rayleigh fading as given by equation 18. Using this equation, the envelope voltage r, is also used to decide the signal present state when the test r, exceeds the threshold level, otherwise the signal of interest is not present. Also, according to the above analysis, the initialisation processor operating condition can be fulfilled by calculating the steady state values of the internal memories of the filter by the direct involvement of the first received samples.

C. The Coherent Limiter: In the practical terms the Optimum Bandwidth filter, should perform the suppression of ISI, above noise, since this is suitable for time-variant signal scenario (Mobile nodes) . This problem can be alleviated by using a coherent *limiter* prior to the phase detector, to limit the incoming signals to an acceptable level. The limiter level must be rather low with respect to the value of the incoming Intersymbol Interference, and precluding for higher noisy signal levels where maximum rejection is required.

D. Equalisation Process: Using IIR equalization filters will ensure the stability and smooth frequency response, and S/N ratio at the equalizer's output. Therefore, the value of S/N is enhanced by whitening the noise, which is affecting the received signal.

### *III. Data observation using initialising Processor*

Samples of the recorded and processed data are illustrated in Figure .2. Representing simulating noise-like signal scenarios. As can be seen from these scenarios, the capability of the ***Processor*** in suppressing the noisy signal consecutively and at different scenarios.

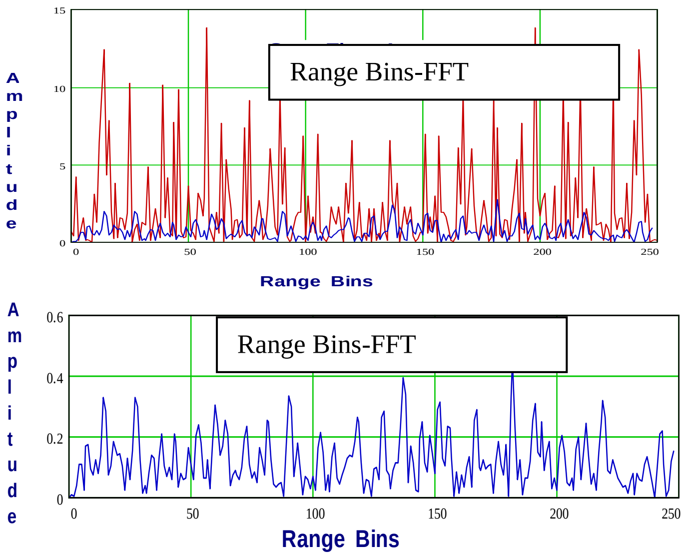


Figure.2. the initialisation Processor after FFT and its effect on noisy signal.

The rejection capability of the initialisation technique used with OFDM system, is demonstrated in Fig.3. Which shows the initialisation output of Gaussian noise, with the effect of initialisation, when the noise-whitening filter is used?

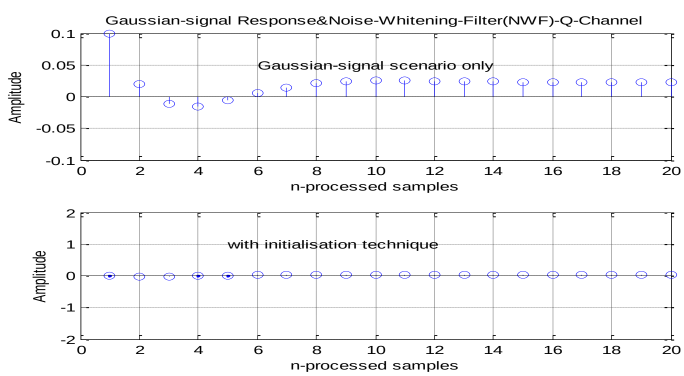


Fig.3. Gaussian noise spectrum rejection with the aid of noise-whitening filter and initialisation technique.

Least square estimator was used, and the output signal constellation is shown in fig.4, from which sixteen 16–QAM different symbol values, as an indication that equalization has been performed on training signal.

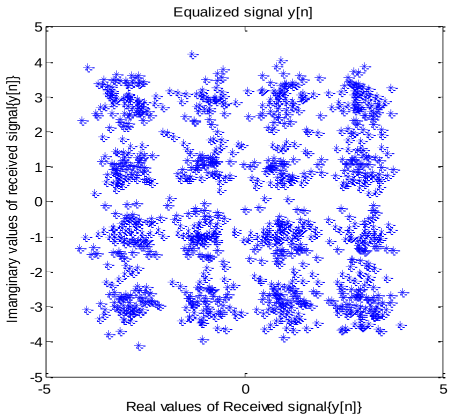


Fig.4. Constellation of 16-QAM signal with equalization

## IV. CONCLUSIONS

The processor was described in this work in terms of the inclusion of an initialisation technique, digital filtering and other auxiliary processing parameters. Based on this model, the equalizer is simplified as it is connected with the initialisation technique, to improve its function. The comparisons between the proposed technique and the low order equalization was evident in a reduction in calculation burden, and working in the transient mode , with limited number of samples, while preserving performance.

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