MTech Project Report

On

Digital Non-Linear SI Cancellation for In-band Full Duplex Radios

For

The Partial Fulfillment of the degree of Master of Technology in Electrical Engineering

2018-2020

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1. Problem Formulation

Full duplex systems require efficient and strong self-interference cancellation in order to suppress the SI below or up to the noise floor and to work properly. A significant part of this Self Interference signal is due to the nonlinear effects caused by various transceiver impairments. As such linear cancellation alone is not usually sufficient and nonlinear cancellation techniques have to be used. That's why a combination of SI cancellation in the analog and in the digital domain is necessary to suppress the SI signal down to the level of the receiver's noise floor.

The context of the problem is to model the nonlinear self-interference signal effectively and suppress it up to or below the receiver's noise level in the digital domain using adaptive filtering techniques.

1. INTRODUCTION

The wireless revolution has turned to be resulted in ever increasing demands on our limited wireless spectrum, forcing us for finding systems/techniques with higher spectral efficiencies. Among various ways to increase the spectral efficiency, in band full duplexer are very prominent. Most contemporary communication devices/systems have terminals that functions as both transmitter and receivers that either operate in half duplex or out of band full duplex (transmit or receive data either at different time or over different frequency bands e.g. LTE). But on the contrary, In-band Full Duplexer (fig 1) is a wireless technology that allows the wireless device to transmit and receive data simultaneously in the same frequency band thereby efficiently increasing the utilization of the spectrum and doubling the spectrum efficiency. But the main drawback of this is its own transmitter that causes strong self-interference at the receiver that saturates the receiver chain and hence has to be effectively cancelled.

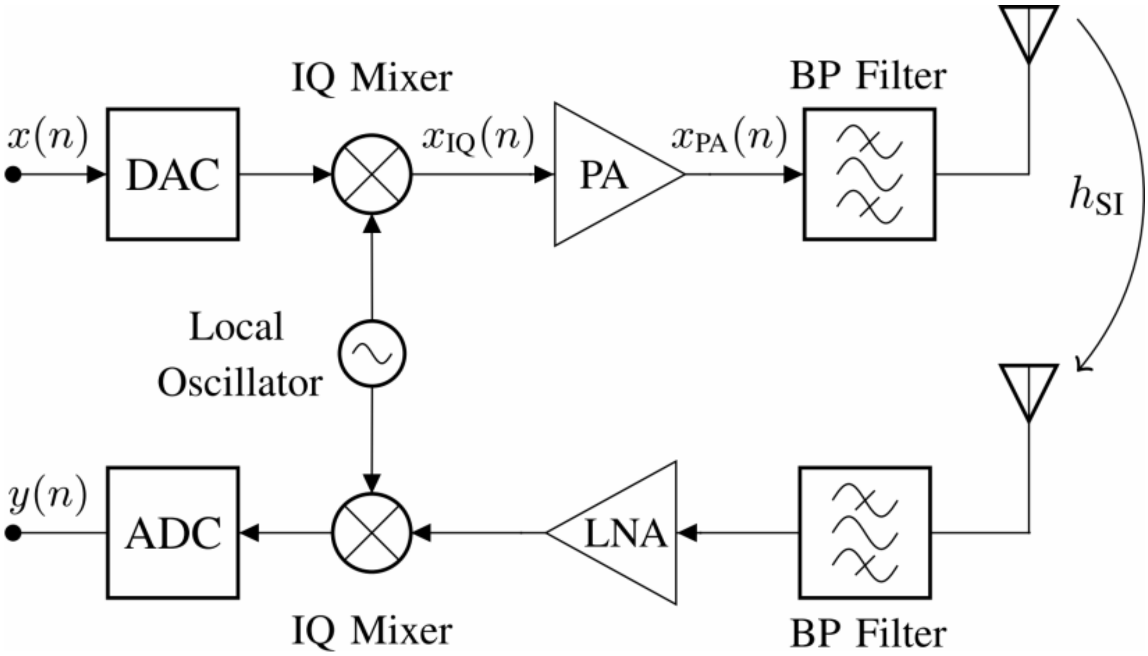

Fig 1: Basic model of a full duplex Transceiver

2. Causes of Self Interference

The received signal of a full duplex system does not only have the required signal but also several copies of the transmitted signal through various paths. All these caused due to the following:

1. Transmitter and receiver coupling at the antenna
2. Leakage of the transmit signal from the power amplifier to the received signal
3. Reflections of the transmitted signal from the external environment picked up by the antenna.

Therefore, due to above mentioned reasons SI arises and needed to be cancelled both in analog and digital domain to suppress it below the noise floor. Analog cancellation can be done in two ways either passive (through physical isolation between the transmitter and the receiver) or active (that is through injection of a cancellation signal). But a residual SI signal still remain present in the receiver after the RF cancellation stage. Residual SI signal should be easily cancellable in the digital domain as it is caused by a signal that is fully known but that's is not the case because of many transceivers' impairments and non-linearities. There is various baseband non linearities present, IQ imbalances, and power amplifier nonlinearities that distorts the signal. Of all these power amplifier (PA) nonlinearities is the bottleneck in full duplex communications and have to be dealt with.

3. Non-Linear Self Interference Channel Modeling

For discrete time non-linear power amplifier modeling Parallel Hammerstein model is used. Let us denote the transmit baseband by x_n. Then the output of the power amplifier is modelled as

$$x_n^{PA} = \sum_{p=1}^{P} \sum_{k=1}^{M-1} f(p,k)\psi_p\big(x(n-k)\big) \qquad [1]$$

where the basis functions are defined as $\psi_p\big(x(n)\big) = |x_n|^{p-1}x_n$(p is odd as even terms fall outside the desired spectrum), $f(p,k)$ are FIR filter impulse responses of the PH branches, M denotes the memory length, and P denotes the non-linearity order of the model.

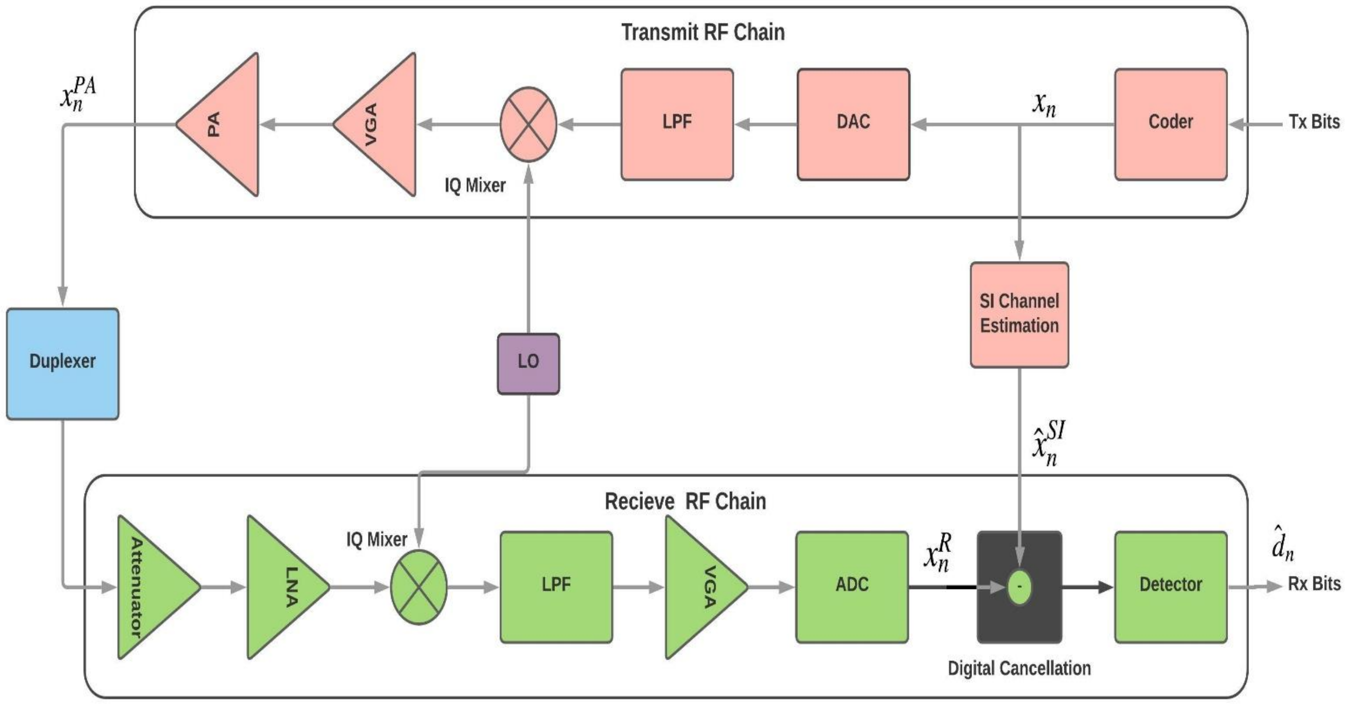

Fig 2: Showing the flow of samples in the FD design

4. Digital SI Cancellation Parameter Estimation

Even though the discussed model is non-linear, it is linear with respect to the parameters $f(p,k)$ that can provide efficient estimation using least squares methods. Now the aim is to estimate these parameters based on the above SI model, regenerate the SI signal and subtract it from the overall received signal at the digital baseband. Block diagram of overall nonlinear SI channel modelling is shown in fig having different branches, corresponding to nonlinear terms of different order.

Now the total received signal before digital cancellation block is given as

$$x_n^R = d_n + w_n + x_n^{SI} \qquad [2]$$

Where d_n is the signal of interest, w_n is additive noise, and x_n^{SI} is the received signal interference Output of the digital SI canceller is

$$\hat{d}_n = x_n^R - \hat{x}_n^{SI} \qquad [3]$$

Where the SI estimate $\hat{x}_n^{SI}$ is given as

$$\hat{x}_n^{SI} = \sum_{p=1}^{P} \sum_{k=1}^{M-1} \hat{f}(p,k)\psi_p\big(x(n-k)\big) \qquad [4]$$

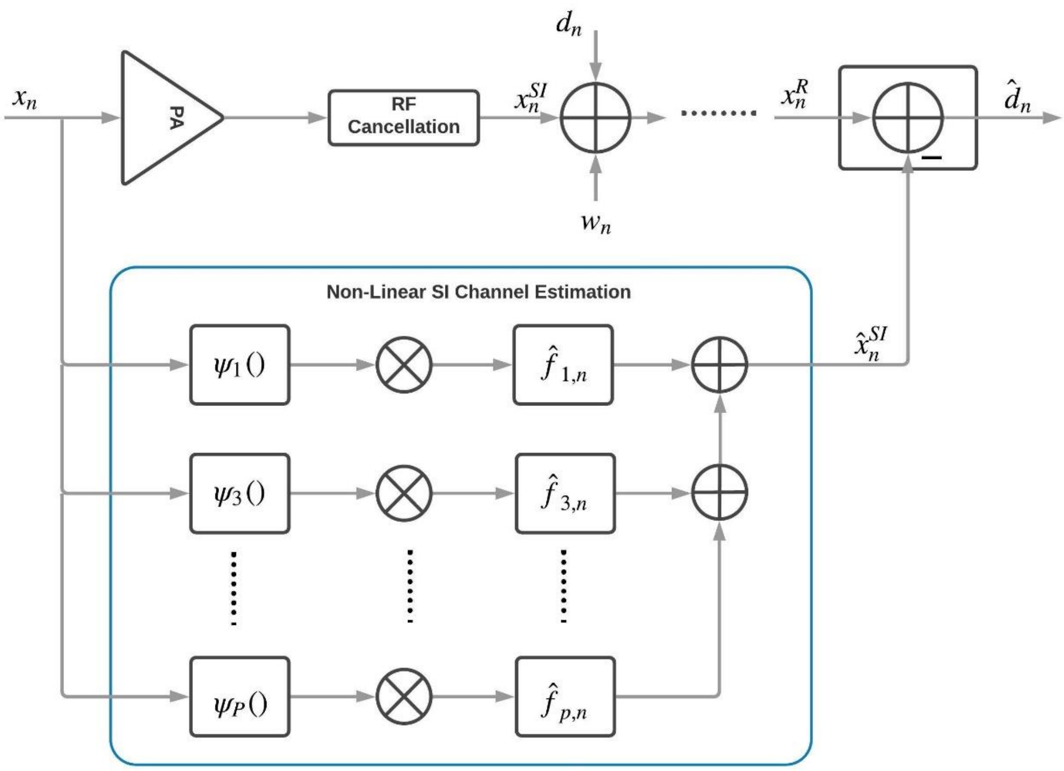

Fig 3: Structure for non-linear SI estimation and cancellation

In order to derive the estimator for $\hat{f}(p,k)$, we proceed with the vector representation of the required formulas with N observed samples of x_n^R

$$\boldsymbol{X^R} = \boldsymbol{\Psi f} + \boldsymbol{d} + \boldsymbol{w}$$

$$\boldsymbol{X^R} \triangleq \left[\boldsymbol{x_n^R} \quad \boldsymbol{x_{n+1}^R} \cdots\cdots \boldsymbol{x_{n+N-1}^R}\right]^{\boldsymbol{T}}$$

$$\boldsymbol{f} \triangleq \left[\boldsymbol{f_{1,0}} \cdots\cdots \boldsymbol{f_{1,M}} \;\; \boldsymbol{f_{3,0}} \cdots\cdots \boldsymbol{f_{3,M}} \;\; \boldsymbol{f_{P,0}} \cdots\cdots \boldsymbol{f_{P,M}}\right]^{\boldsymbol{T}}$$

$$\boldsymbol{\Psi} \triangleq [\boldsymbol{\psi_1} \quad \boldsymbol{\psi_3} \cdots\cdots \boldsymbol{\psi_P}]$$

$$\boldsymbol{\psi_P} = \begin{bmatrix} \boldsymbol{\Psi_P(n)} & \boldsymbol{\Psi_P(n-1)} & \cdots & \boldsymbol{\Psi_P(n-M)} \\ \boldsymbol{\Psi_P(n+1)} & \boldsymbol{\Psi_P(n)} & \cdots & \boldsymbol{\Psi_P(n-M+1)} \\ \vdots & \vdots & \ddots & \vdots \\ \boldsymbol{\Psi_P(n+N-1)} & \boldsymbol{\Psi_P(n+N-2)} & \cdots & \boldsymbol{\Psi_P(n-M+N-1)} \end{bmatrix}$$

Now the least square estimate is derived by as the vector $\boldsymbol{f}$ which minimizes the power of the digital SI canceller output $\hat{d}_n$

Now the least square estimate is given as

$$\hat{f}^{LS} = \arg\min_f \left\| X^R - \hat{X}^{SI} \right\|^2 = \arg\min_f \left\| X^R - \boldsymbol{\Psi}\hat{f} \right\|^2$$

Now to avoid the matrix inversion step we avoid the least square estimator design and resort to linear adaptive filtering techniques and will use LMS filter for the estimation of filter coefficients in our cancellation design. Details for LMS filter and cancellation design will be discussed and shown in the subsequent sections.

5. LMS Filter

Least Mean Square filters are class of adaptive filters that mimics a desired filter by finding the filter coefficients that relate to producing the least mean square of the error signal (difference between the desired signal and input/actual signal). It's a stochastic gradient based method of steepest descent. LMS algorithm uses the estimates of the gradient vector from the current data by incorporating an iterative method that successively updates the weight vector that gives the least mean square error. One of the main advantages of LMS is that it does not require correlation calculation and matrix inversion that makes the process efficient.

The basic concept in the LMS filter is that it approaches the optimum filter weights by updating the filter weights in a manner that it converges to the optimum filter weight. It functions by assuming small weight and at each step weights are updated according to the gradient of the mean square error. The mean square as function of filter weights is quadratic in nature and thus have only one extremum that minimizes the error toward the optimal weights by ascending or descending down the MSE curve.

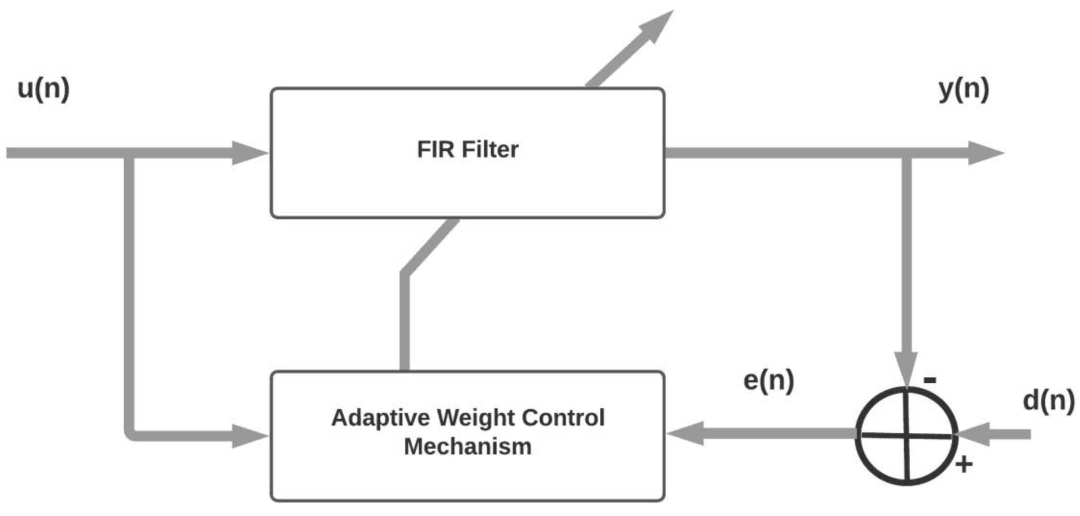

Fig 4: LMS Filter Structure

LMS algorithm for a filter order p is given as:

Parameters: p = filter order

n = no. of samples

μ = step size

Initialization: $\hat{h}$ = zeros(p)

Computation: for n = 0, 1, 2, …….

X(n) = $[x(n), x(n-1), \ldots \ldots \ldots, x(n-p+1)]$

$$e(n) = d(n) - \hat{h}^H(n)x(n)$$

Tap update: $\hat{h}(n+1) = \hat{h}(n) + \mu e^*(n)x(n)$

The two figures shown below are the blocks of LMS filter one being FIR filter and the other one weight control block respectively. In the figure their structure and functioning are shown clearly.

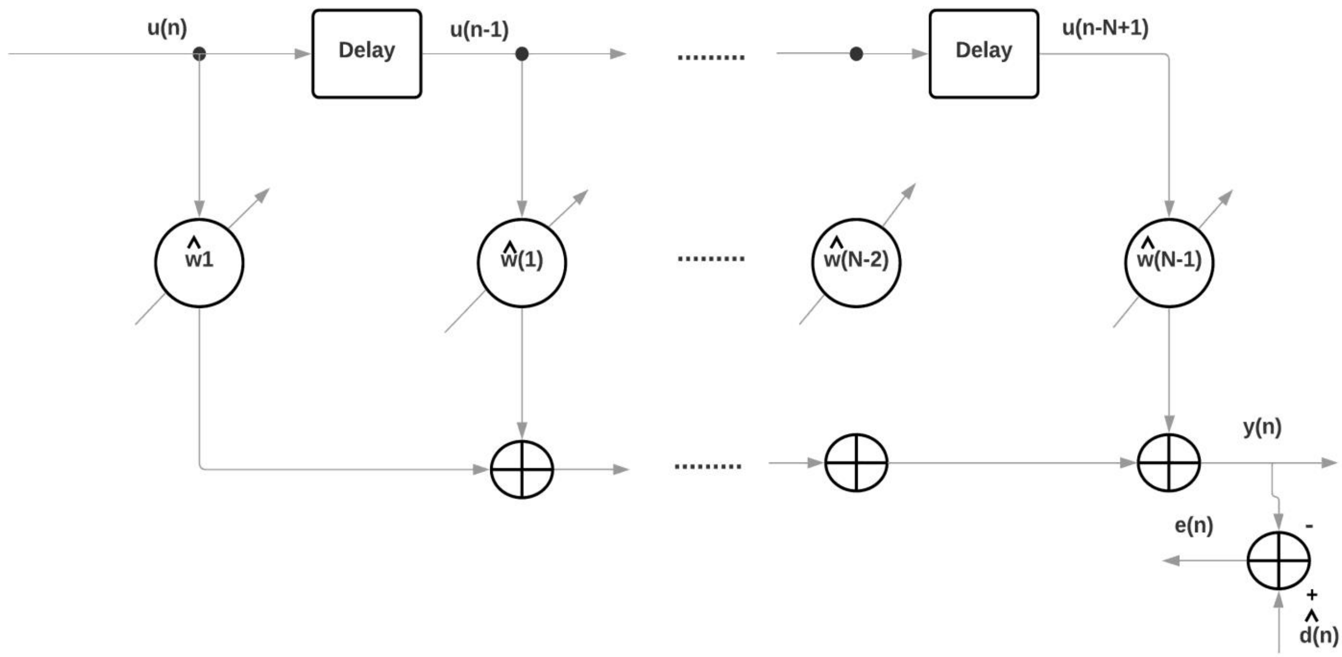

Fig 5: FIR Filter Structure

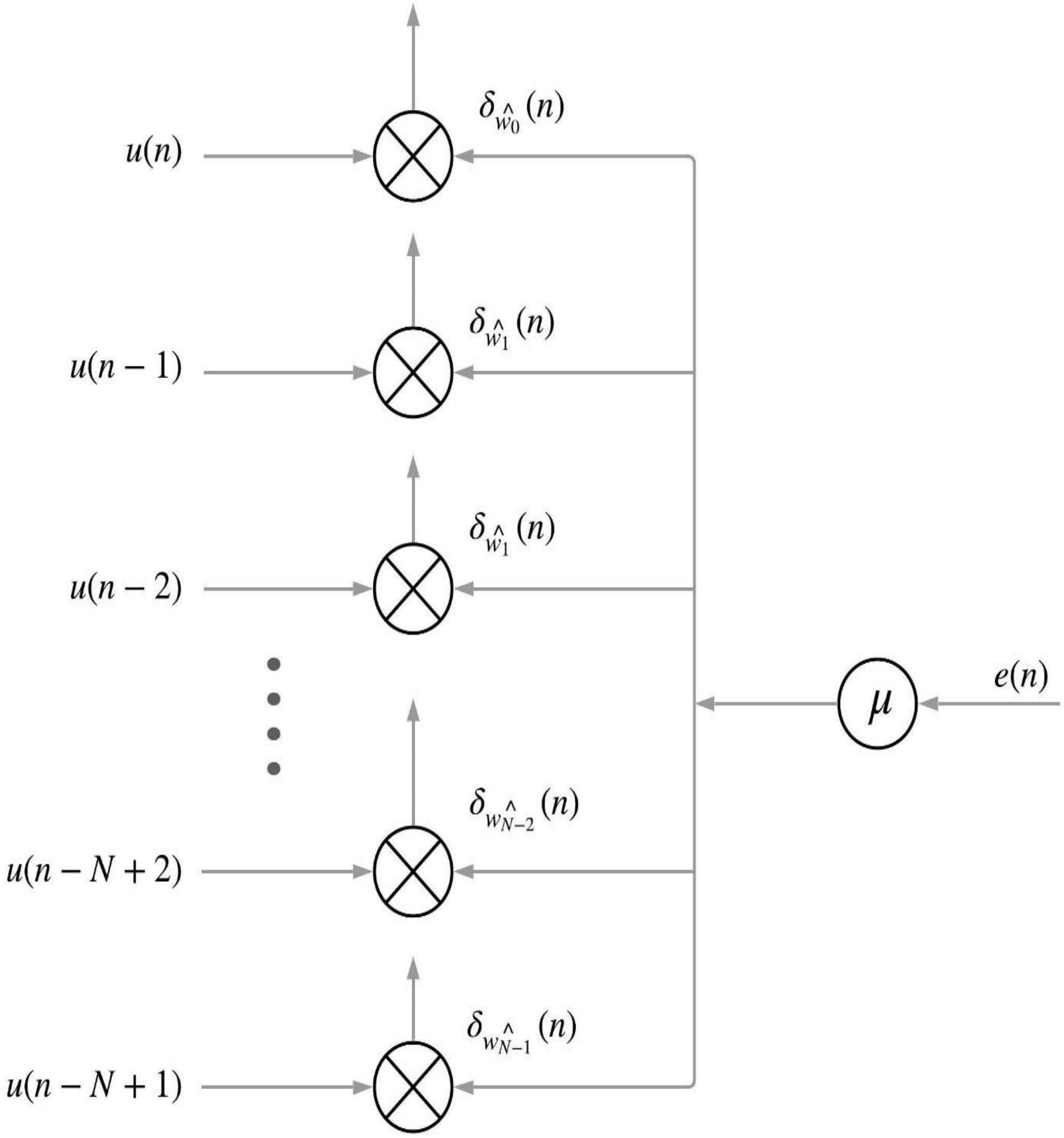

Fig 6: Adaptive Weight Control Mechanism

During the filtering desired signa d(n) supplied to the filter and with the tap input vector (u(n) in this case). This input vector is used in the FIR filter that has an output y(n) that is used for the evaluation of desired signal. After that e(n) is generated to take the modification between the actual needed response and the actual filter outputs shown in the fig 5. In the weight control block tap input vector u(n-k) and the inner product of the estimation error e(n) is calculated as shown in the fig 6.

This setup of LMS filter is used if the signal is linear as it's a linear adaptive filter. But in our FD setup we are not getting a linear signal to be operated on LMS filter for cancellation because of the nonlinearities caused by the power amplifier during transmission. So, we will be modelling these nonlinearities as discussed in the previous sections and according to that model our LMS setup will look like as shown in the figure in the next page.

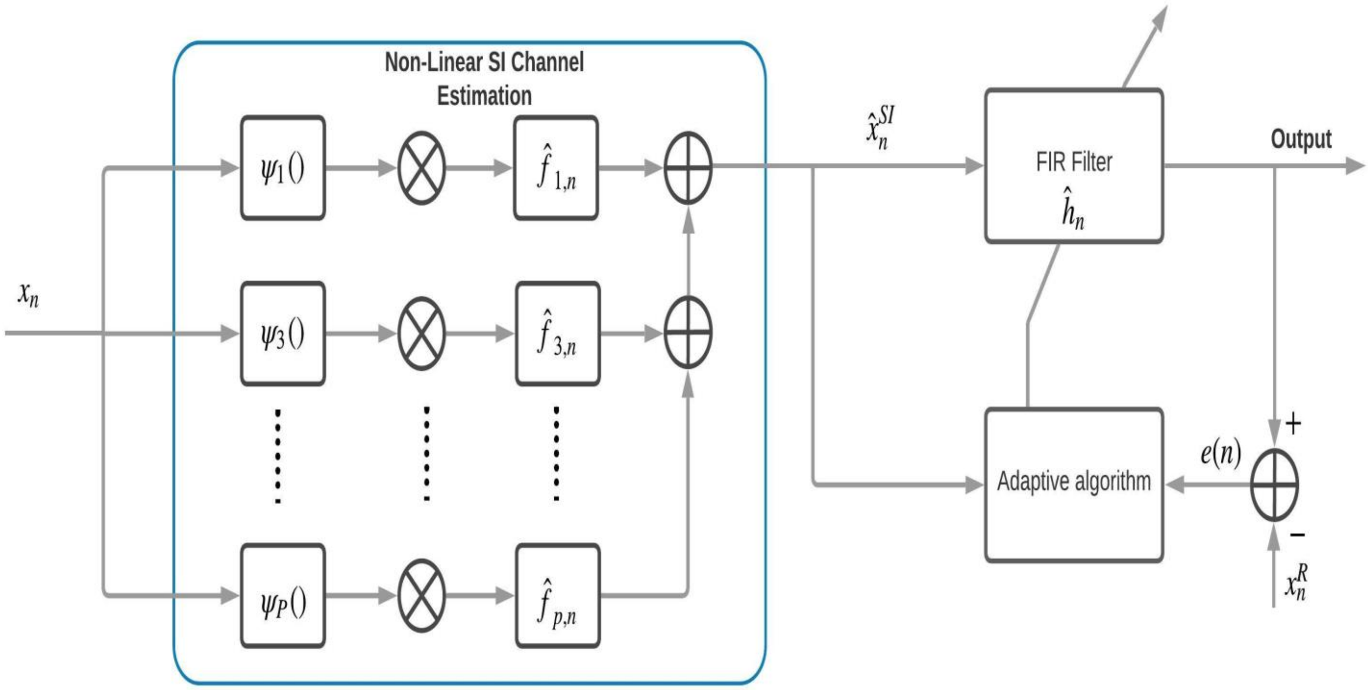

Fig 7: LMS Filter with non-linear SI channel estimation

6. Hardware Setup and Explanation

Following figure shows how the setup is working between the AD9371 transceiver, ZCU102 board and the RF duplexer in the in-band FD system and how they are communicating. FD system setup has the following specifications:

1. Tx power – 30dBm
2. Rx power – according to the 4G standards
3. Bandwidth – 20 MHz
4. Centre Frequency – 2.35 GHZ

5. Setup is running at 61.44 Msamples/sec

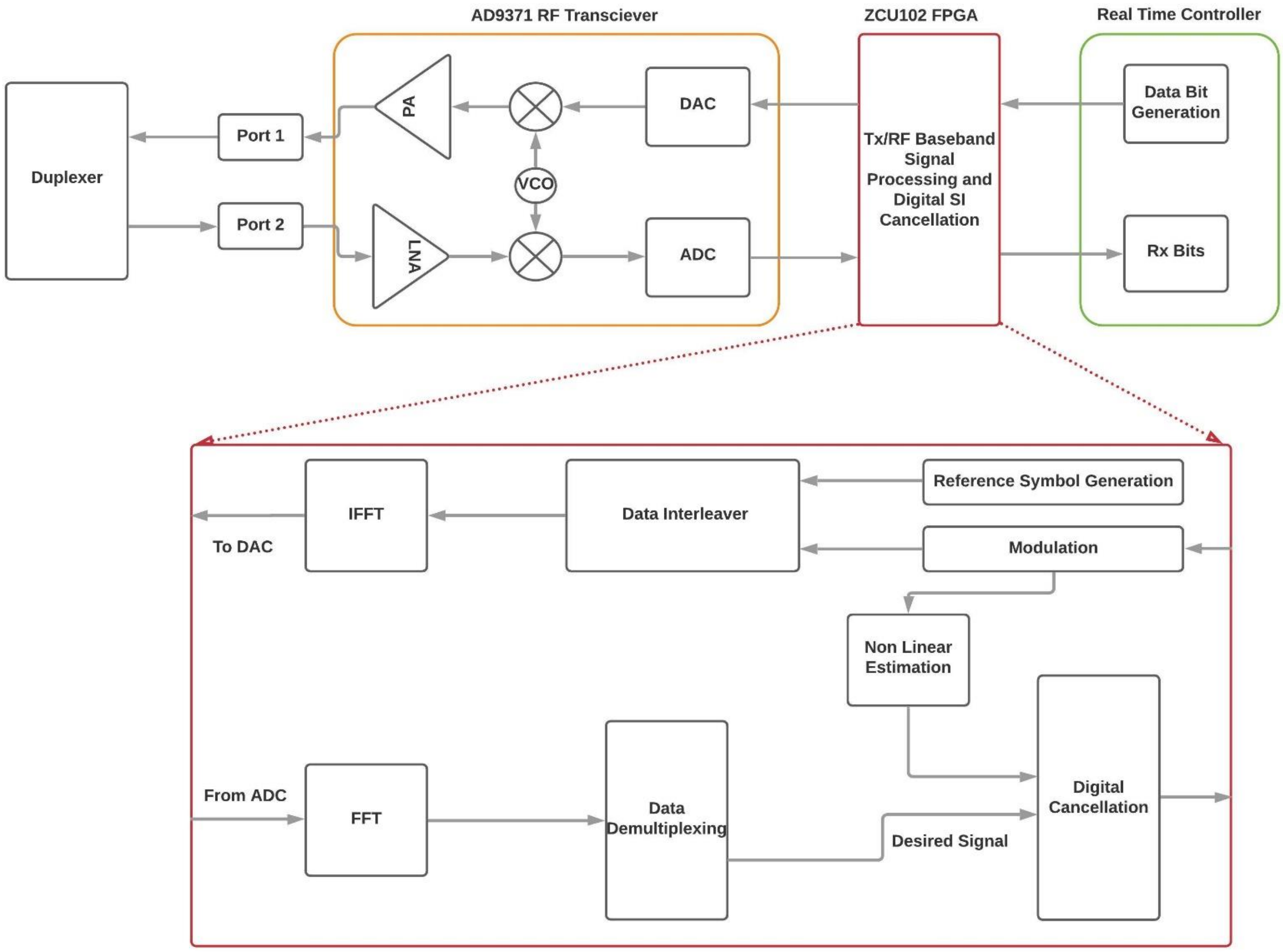

Fig 8: Hardware setup for the FD system

RT Controller: This block is nothing but Linux PC system through which we control the whole setup and provide the OFDM signal (to be transmitted) through II Oscilloscope application where we also see the desired results.

AD9371 Transceiver: AD9371 is a highly integrated RF transceiver equipped with digital to analog converter 10-bit accuracy and analog to digital converter with 12-bit accuracy is configured for the FD system.

ZCU102 Board: Bootable SD card with AD9371 Linux image is prepared to configure the ZCU102 FPGA for controlling the SPI from RT controller and running the HDL design on the FPGA.

Duplexer: Key function of this is to keep transmit energy out of the receiver to prevent desensitization but it is not completely able to isolate the Tx and Rx and therefore some part of Tx interferes with the Rx and thus contributing the SI in the system needed to be eliminated.

Transmission: A 5 MHz OFDM signal is generated in the RT controller for the transmission. The bits generated are sent to the FPGA for the processing where after the modulation data symbols are interleaved with the reference signal and stored in the look up tables. After that processed signal goes to the AD9371 transceiver block where it gets interpolated and fed to the digital to analog converter for the transmission.

Digital SI Cancellation: The main aim of this cancellation is to suppress the residual self-interference signal remaining after the analog SI cancellation. Digital self-interference cancellation is mainly the reconstruction of self-interference and subtracting it from the received signal containing the signal of interest. In order to design a real time digital canceller filter, we use FPGA implementation using Vivado software and AD9371 HDL design platform.

LMS filter: Channel estimation for Full Duplex has estimations for the TX and Rx ports in its full duplex nodes by rebuilding the interference using the known Tx data. For channel estimation PH model is being used that is discussed in the above sections. For the cancellation structure using the channel estimation model LMS filter IP is built in HLS Vivado that uses the baseband samples of the transmitted data to rebuild self-interference in the digital domain and subtract it from the received signal. Below synthesis report for the LMS filter IP designed in Vivado HLS is shown that will be incorporated in the design chain.

Performance Estimates

Timing (ns)

Summary

Clock	Target	Estimated	Uncertainty
ap_clk	16.00	13.145	2.00

Latency (clock cycles)

Summary

Latency		Interval		
min	max	min	max	Type
2	2	1	1	function

Detail

Instance

N/A

Loop

N/A

Utilization Estimates

Summary

Name	BRAM_18K	DSP48E	FF	LUT	URAM
DSP	-	11	-	-	-
Expression	-	54	0	2042	-
FIFO	-	-	-	-	-
Instance	-	-	-	-	-
Memory	-	-	-	-	-
Multiplexer	-	-	-	264	-
Register	-	-	18981	-	-
Total	0	65	18981	2306	0
Available	624	1728	460800	230400	96
Utilization (%)	0	3	4	1	0

Detail

Instance

N/A

Justification for the Flip-Flops utilization: In the C code of our LMS filter we have used a delay of 191 samples between Tx and Rx and to accommodate for that we use 'for' loop that iterate for 191 times for three streams and each sample is of 16 bit size so utilization is as follows 191*16*6=18336. Along with these two more arrays of size one with samples of bit size 22 and 16 respectively also used so for that 22*6=132 and 16*6=96. Total flip-flop count is 18336+132+96=18564. This explains the most of the flip-flop utilization in our design.

8.Result

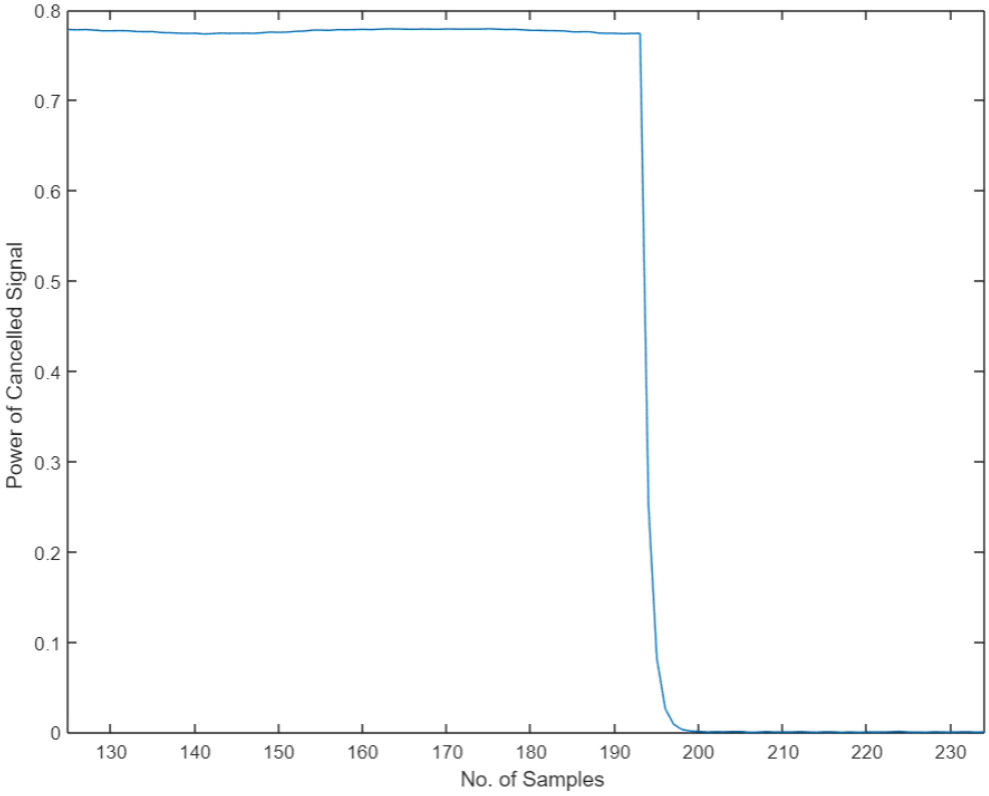

Fig 9: Response of Designed LMS Filter for Non-Linear SI Cancellation

- From the time domain response of the LMS filter we can easily observe that power of the cancelled signal quite high up to 195 samples but after 195 samples power of the cancelled signal gradually decreases and becomes zero that is quite justified as there is delay of 191 samples between Tx and Rx in our FD design.
- With the same design of LMS filter we got a cancellation of 34 dB with only the power amplifier in the transmission path that we checked on MATLAB.

References

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