FPGA IMPLEMENTATION OF LOW-COMPLEXITY ICA BASED BLIND MULTIPLE-INPUT MULTIPLE-OUTPUT OFDM RECEIVERS

A THESIS SUBMITTED IN PARTIAL FULFILLMENT OF THE
REQUIREMENTS FOR THE DEGREE OF

Master of Technology

In

VLSI DESIGN & EMBEDDED SYSTEM

By

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Roll No: 207EC209

DEPARTMENT OF ELECTRONICS & COMMUNICATION ENGINEERING

NATIONAL INSTITUTE OF TECHNOLOGY

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C E R T I F I C A T E

This is to certify that the thesis entitled, **"FPGA IMPLIMENTATION OF LOW-COMPLEXITY ICA BASED BLIND MULTIPLE-INPUT MULTIPLE-OUTPUT OFDM RECEIVERS"** submitted by Mr. **VENKATA SUBBA REDDY.B (207EC209)**in partial fulfillment of the requirements for the award of Master of Technology Degree in **ELECTRONICS & COMMUNICATION ENGINEERING** with specialization in **"VLSI DESIGN & EMBEDED SYSTEM"** at the National Institute of Technology, Rourkela is an authentic work carried out by him under my supervision and guidance.

To the best of my knowledge, the matter embodied in the thesis has not been submitted to any other University / Institute for the award of any Degree or Diploma.

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CONTENTS

ABSTRACT

Now-a-days the problem with almost all of the wireless communication systems is multi-path fading channels. Orthogonal Frequency Division Multiplexing (OFDM) has become a popular modulation method in high speed wireless communications. By partitioning a wideband fading channel into flat narrowband channels, OFDM is able to mitigate the effects of multi-path fading using an equalizer. In a typical OFDM broadband wireless communication system, a guard interval is inserted to avoid the inter-symbol interference (ISI) and the inter-carrier interference (ICI). This guard interval is required to be at least equal to the maximum channel delay spread. Otherwise equalization is required at the receiver.

To meet the ever growing demand for higher data rates in wireless communication systems, multiple transmit and receive antennas can be employed to make use of the spatial diversity by transmitting data in parallel streams. Such spatial multiplexing Multiple-Input Multiple-Output (MIMO) systems have been shown to obtain significantly higher data rates than Single-Input Single-Output (SISO) systems. This increase in data rate can be achieved without the need of additional bandwidth or transmit power, provided that sufficient multipath diversity is present.

In this thesis Independent Component Analysis (ICA) based methods are used for blind detection in MIMO systems. ICA relies on higher order statistics (HOS) to recover the transmitted streams from the received mixture. Blind separation of the mixture is achieved based on the assumption of mutual statistical independence of the source streams. The use of HOS makes ICA methods less sensitive to Gaussian noise. ICA increase the spectral efficiency compared to conventional systems, without any training/pilot data required.

ICA is usually used for blind source separation (BSS) from their mixtures by measuring non-Gaussianity using Kurtosis. Many scientific problems require FP arithmetic with high precision in their calculations. Moreover a large dynamic range of numbers is necessary for signal processing. FP arithmetic has the ability to automatically scale numbers and allows numbers to be represented in a wider range than fixed-point arithmetic. Nevertheless, FP algorithm is difficult to implement on the FPGA, because the algorithm is so complex that the area (logic elements) of FPGA leads to excessive consumption when implemented. A simplified 32-bit FP implementation includes adder, Subtractor, multiplier, divider, and square rooter The FPGA design is based on a hierarchical concept, and the experimental results of the design are presented.

LIST OF FIGURES

LIST OF TALBES

LIST OF ABBREVIATION

ADSL	Asymmetric Digital subscriber Line
AMPS	Advanced Mobile Phone Service
AWGN	Additive White Gaussian Noise
BER	Bit Error Rate
BPSK	Binary Phase Shift Keying
BSS	Blind Source Separation
CDMA	Code Division Multiple Access
CMA	Constant modulus algorithm
CP	Cyclic Prefix
CSI	Channel State Information
DAB	Digital Audio Broadcast
DFT	Discrete Fourier Transform
DSL	Digital Subscriber Lines
DSP	Digital Signal Processer
DVB	Digital Video Broadcast
DVD	Digital Video Display
ECG	Electrocardiogram
EEG	Electroencephalograms
FDE	Frequency Domain Equalizer
FDM	Frequency Division Multiplexing
FDMA	Frequency Division Multiple Access
FM	Frequency Modulation
FP	Floating-point
FPGA	Field Programmable Gate Array Logic
FSK	Frequency Shift keying
GMSK	Gaussian Minimum Shift Keying
GPRS	Global Packet Radio Service
GSM	Global System for Mobile communications
HDSL	High-bit-rate Digital Subscriber Lines
HOS	Higher Order Statistics
IBI	Inter Block Interference
ICI	Inter Carrier Interference

IDFT	Inverse Discrete Fourier Transform
IEEE	Institute of Electrical and Electronics Engineers
IFFT	Inverse Fast Fourier Transform
IIR	Infinite Impulse Response
IMD	Inter- Modulation Distortion
ISO	International Standards Organization
ISI	Inter Symbol Interference
LAN	Local Area Network
LTI	Linear Time Invariant
MC	Multicarrier Communication
MAI	Multiple Access Interference
MCM	Multi carrier Modulation
MEG	Magneto Encephalograms
MIMO	Multi Input Multi Output
ML	Maximum Likelihood
MMSE	Minimum Mean Squared Error
NMT	Nordic Mobile Telephone
NTT	Nippon Telephone and Telegraph
OFDM	Orthogonal Frequency Division Multiplexing
OSI	Open Systems Interconnection
P/S	Parallel to Serial
PAPR	Peak-to-Average Power Ratio
QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying
RF	Radio Frequency
SISO	Single Input Single Output
SNR	Signal to Noise Ratio
SOS	Second Order Statistics
S/P	Serial to Parallel
SRT	Sweeney, Robertson, and Tocher
TACS	Total Access Communications system
TCP	Transmission Control Protocol
TDE	Time Domain Equalizer
TDMA	Time Division Multiple Access

VoIP	Voice over IP
VDSL	Very-high-speed Digital Subscriber Lines
VHDL	Very High Speed Hardware Descriptive Language
VLSI	Very Large Scale Integrated Circuits
WLAN	wireless Local Area Network
ZF	Zero Forcing
ZP	ZeroPadding

CHAPTER 1

INTRODUCTION

1.1 Introduction to frequency division multiplexing

The demand for high data rate services has been increasing very rapidly and there is no slowdown in sight. Almost every existing physical medium capable of supporting broadband data transmission to our homes, offices and schools are being can be used in the future. Same of the technology includes (Digital Subscriber Lines, Cable Modems, and Power Lines in wired and wireless media. Often, these services require very reliable data transmission over very harsh environments. Most of these transmission systems experience degradation in form of attenuation, noise, multipathinterference, the technology, non-linearity, and must meet many constraints, such as finite transmit power and most importantly should be supported and double cost. One physical-layer technique that has recently gained high popularity due to its robustness in dealing with these impairments is multi-carrier modulation.

Orthogonal Frequency Division Multiplexing (OFDM) is a special form of multi carrier modulation technique where signals are used over narrowband channel, mutually orthogonal. Due to the recent advancements in digital signal processing (DSP) and very large-scale integrated circuits (VLSI) technologies, many of the initial obstacles of OFDM implementations have been overcome.

The major advantages of OFDM are its ability to convert a frequency selective fading channel into several nearly flat fading channels hence providing high spectral efficiency. However, one of the main disadvantages of OFDM is its sensitivity against carrier frequency offset which causes attenuation and rotation of subcarriers termed as intercarrier interference (ICI)[1,2]. The undesired ICI degrades the performance of the system.

Attraction towards OFDM is it's due to its way of handling the multipath interference at the receiver. Multipath phenomenon generates two effects (a) Frequency selective fading and (b) Intersymbol interference (ISI). On the other hand, modulating symbols at a very low rate makes the symbols much longer than channel impulse response and hence reduces the ISI. Insertion of an extra guard interval between consecutive OFDM symbols can reduce the effects of ISI even more but leads to low system capacity. Among them, the multiple input–multiple output (MIMO) system using multiple antennas at both the transmitter and the receiver has attracted a lot of research interest due to its potential to increase the system capacity without extra bandwidth .

To combat the effect of frequency selective fading, MIMO is generally combined with orthogonal frequency-division multiplexing (OFDM) technique, which transforms the frequency-selective fading channels into parallel flat fading sub channels, as long as the cyclic prefix (CP) inserted at the beginning of each OFDM symbol is longer than or equal to the channel length [5]. In this case, the signals on each subcarrier can be easily detected by a one-tap frequency domain equalizer (FDE). In cases where a short CP is inserted for increasing bandwidth efficiency, or because of some unforeseen channel behavior, the effect of frequency-selective fading cannot be completely eliminated, and intercarrier interference (ICI) and intersymbol interference (ISI) will be introduce. In this case, the signals on each subcarrier can be easily detected by a one-tap time domain equalizer (TDE). Equalization is important in MIMO-OFDM systems.

1.2 History of mobile wireless communications

The history of mobile communication [3, 4] can be categorized into 3 periods:

* The pioneer era
* The pre-cellular era
* The cellular era

In the pioneer era, a great deal of the fundamental research and development in the field of wireless communications took place. The postulates of electromagnetic (EM) waves by James Clark Maxwell during the 1860s in England, the demonstration of the existence of these waves by Heinrich Rudolf Hertz in 1880s in Germany and the invention and first demonstration of wireless telegraphy by Guglielmo Marconi during the 1890s in Italy were representative examples from Europe. Moreover, in Japan, the Radio Telegraph Research Division was established as a part of the Electro technical Laboratory at the Ministry of Communications and started to research wireless telegraph in 1896.

From the fundamental research and the resultant developments in wireless telegraphy, the application of wireless telegraphy to mobile communication systems started from the 1920s. This period, which is called the pre-cellular era, began with the first land-based mobile wireless telephone system installed in 1921 by the Detroit Police Department to dispatch patrol cars, followed in 1932 by the New York City Police Department. These systems were operated in the 2MHz frequency band.

In 1946, the first commercial mobile telephone system, operated in the 150MHz frequency band, was set up by Bell Telephone Laboratories in St. Louis. The demonstration system was a simple analog communication system with a manually operated telephone exchange. Subsequently, in 1969, a mobile duplex communication system was realized in the

450MHz frequency band. The telephone exchange of this modified system was operated automatically.

The cellular zone concept was developed to overcome this problem by using the propagation characteristics of radio waves. The cellular zone concept divided a large coverage area into many smaller zones. A frequency channel in one cellular zone is used in another cellular zone. However, the distance between the cellular zones that use the same frequency channels is sufficiently long to ensure that the probability of interference is quite low. The use of the new cellular zone concept launched the third era, known as the cellular era.

So far, the evolution of the analog cellular mobile communication system is described. There were many problems and issues, for example, the incompatibility of the various systems in each country or region, which precluded roaming. In addition, analog mobile communication systems were unable to ensure sufficient capacity for the increasing number of users, and the speech quality was not good.

To solve these problems, the R&D of cellular mobile communication systems based on digital radio transmission schemes was initiated. These new mobile communication systems became known as the second generation (2G) of mobile communication systems, and the analog cellular era is regarded as the first generation (1G) of mobile communication systems [5,6].

1G analog cellular system was actually a hybrid of analog voice channels and digital control channels. The analog voice channels typically used Frequency Modulation (FM) and the digital control channels used simple Frequency Shift keying (FSK) modulation. The first commercial analog cellular systems include Nippon Telephone and Telegraph (NTT) Cellular in Japan, Advanced Mobile Phone Service (AMPS) in US, Australia, China, Southeast Asia, Total Access Communications system (TACS) in UK, and Nordic Mobile Telephone (NMT) in Norway, and Europe.

2G digital systems used digital radio channels for both voice (digital voice) and digital control channels. 2G digital systems typically use more efficient modulation technologies. Which include 2-level Gaussian Minimum Shift Keying (GMSK) for GSM.

Digital radio channels offer a universal data transmission system, which can be divided into many logical channels that can perform different services. 2G also uses multiple access (or multiplexing) technologies to allow more customers to share individual radio channels or use narrow channels to allow more radio channels into a limited amount of radio spectrum band. The 3 basic types of access technologies used in 2G are: frequency division multiple accesses (FDMA), time division multiple access (TDMA), and code division multiple access (CDMA). The technologies reduce the RF channel bandwidth (FDMA), share a radio channel by assigning users to brief timeslot (TDMA), or divide a wide RF channel into many different coded channels (CDMA).Improvements in modulation techniques and multiple access technologies amongst other technologies inadvertently led to 2.5G and 3G. For example, EDGE can achieve max 474 kbps by using 8-PSK with the existing GMSK. This is 3x more data transfer than GPRS.

1.3 Motivation

Orthogonal Frequency Division Multiplexing (OFDM) has recently gained fair degree of prominence among modulation schemes due to its intrinsic robustness to frequency selective Multipath fading channels. OFDM system also provides higher spectrum efficiency and supports high data rate transmission. This is one of the main reasons to select OFDM a candidate for systems such as Digital Audio Broadcasting (DAB), Digital Video Broadcasting (DVB), Digital Subscriber Lines (DSL), and Wireless local area networks (HiperLAN/2) and in IEEE 802.11a, IEEE 802.11g.

The focus of future fourth-generation (4G) mobile systems is on supporting high data rate services such as deployment of multi-media applications which involve voice, data, pictures, and video over the wireless networks. At this moment, the data rate envisioned for 4G networks is 1 GB/s for indoor and 100Mb/s for outdoor environments. Orthogonal frequency division multiplexing (OFDM) is a promising candidate for 4G systems because of its robustness to the multipath environment. One of the main disadvantages of OFDM is its sensitivity against carrier frequency offset which causes intercarrier interference (ICI). The undesired ICI degrades the performance of the system.

It is not possible to make reliable data decisions unless the ICI powers of OFDM systems are minimized. Thus, an accurate and efficient Intercarrier Interference (ICI) reduction procedure is necessary to demodulate the received data.

1.4 Literature survey

It is well known that Chang proposed the original OFDM principles in 1966, and successfully achieved a patent in January of 1970. OFDM is a technique for transmitting data in parallel by using a large number of modulated sub-carriers. These sub-carriers divide the available bandwidth and are sufficiently separated in frequency so that they are orthogonal. The orthogonality of the carriers means that each carrier has an integer number of cycles over a symbol period.

In 1971, Weinstein and Ebert proposed a modified OFDM system [7] in which the discrete Fourier Transform (DFT) was applied to generate the orthogonal subcarriers waveforms instead of the banks of sinusoidal generators. Their scheme reduced the implementation complexity significantly, by making use of the inverse DFT (IDFT) modules and the digital-to-analog converters. In their proposed model, baseband signals were modulated by the IDFT in the transmitter and then demodulated by DFT in the receiver. Therefore, all the subcarriers were overlapped with others in the frequency domain, while the DFT modulation still assures their orthogonality.

Cyclic prefix (CP) or cyclic extension was first introduced by Peled and Ruiz in 1980 [8] for OFDM systems. In their scheme, conventional null guard interval is substituted by cyclic extension for fully-loaded OFDM modulation. As a result, the orthogonality among the subcarriers was guaranteed. With the trade-off of the transmitting energy efficiency, this new scheme can result in a phenomenal ISI (Inter Symbol Interference) reduction. Hence it has been adopted by the current IEEE standards. In 1980, Hirosaki introduced an equalization algorithm to suppress both inter symbol interference (ISI) and ICI [9], which may have resulted from a channel distortion, synchronization error, or phase error. In the meantime, Hirosaki also applied QAM modulation, pilot tone, and trellis coding techniques in his high-speed OFDM system, which operated in voice-band spectrum.

In 1985, Cimini introduced a pilot-based method to reduce the interference emanating from the multipath and co-channels [10]. In the 1990s, OFDM systems have been exploited for high data rate communications. In the IEEE 802.11 standard, the carrier frequency can go up as high as 2.4 GHz or 5 GHz. Researchers tend to pursue OFDM operating at even much higher frequencies nowadays. For example, the IEEE 802.16 standard proposes yet higher carrier frequencies ranging from 10 GHz to 60 GHz.

However, one of the main disadvantages of OFDM is its sensitivity against carrier frequency offset which causes intercarrier interference (ICI) . The undesired ICI degrades the performance of the system. Numbers of authors have suggested different methods for ICI reduction. These methods are investigated in this thesis and their performances are evaluated.

1.5 Contribution of thesis:

Simulation of MIMO-OFDM was done with different modulation techniques using DFT and IDFT techniques. The digital modulation schemes such as BPSK and QPSK were selected to assess the performance of the designed OFDM system by finding their bit error rate (BER) for different values of signal to noise ratio (SNR).

A blind equalizer is designed to completely suppress both intercarrier interference (ICI) and intersymbol interference (ISI) using second-order statistics of the shifted received OFDM symbols**.** This technique is applied for different number of shifts in the received symbols. The whole simulation is done for both the cyclic prefix (CP) and zero padding (ZP) techniques and compared their performance by finding their Bit Error rate for different values of SNR.

Simulation of ICA was done with floating point algorithms using adders, subs tractors, multipliers, dividers, and square root algorithms

1.6 Thesis outline:

Following the introduction, the rest of the thesis is organized as follows.

In ***Chapter 2***, details regarding Orthogonal Frequency Division Multiplexing, Evolution of ofdm generation and reception Inter-symbol interference, Affect of AWGN on OFDM Advantages, Disadvantages and Applications of OFDM

In C***hapter 3***, details of MIMO-OFDM and the signal model in MIMO-OFDM

In ***Chapter 4,*** details of ICA Based Blind Multiple-Input Multiple-Output OFDM and Signal model in ICA Based Blind MIMO-OFDM.

In***Chapter5,*** details of FPGA implementation of ICA. Algorithm of FastICA, Implementation of floating-point arithmetic

In ***Chapter6,*** details the Results of the thesis

In ***Chapter7,*** details the conclusion and future work

CHAPTER 2

ORTHOGONAL FREQUENCY DIVISION MULTIPLEXING

2.1 OFDM Introduction

Orthogonal Frequency Division Multiplexing (OFDM) has grown to be the most popular communications systems in high speed communications in the last decade. In fact, it has been accepted as the future of wireless communications.

Orthogonal Frequency Division Multiplexing (OFDM) is a multicarrier transmission technique, which utilize the bandwidth available many carriers; each which modulated by a low rate data stream [20, 21]. In term of multiple access technique, OFDM is similar to FDMA in that the multiple user access is achieved by subdividing the available bandwidth into multiple channels that are then allocated to users. However, OFDM uses the spectrum much more efficiently by spacing the channels much closer together. This is achieved by making all the carriers orthogonal to one another, which prevent interference between the closely spaced carriers. This helps in reducing the spacing between carriers

Late 1997, Lucent and NTT submitted proposals to the IEEE for a high speed wireless standard for local area networks (LAN). Eventually, the two companies combined their proposals and it was accepted as a draft standard in 1998 as a standard now known as IEEE802.11a which was adopted in 1999.

2.2 Evolution of ofdm

The evolution of OFDM can be divided into three parts [22]. These are Frequency Division Multiplexing (FDM), Multicarrier Communication (MC) and Orthogonal Frequency Division Multiplexing.

2.2.1 Frequency Division Multiplexing (FDM)

Frequency Division Multiplexing (FDM) has been used for a long time to carry more than one signal over a telephone line. FDM is the concept of using different frequency channels to carry the information of different users. Each channel is identified by the center frequency of transmission.

To ensure that the signal of one channel do not overlap with the signal from an adjacent one, some gap or guard band is left between different channels. Obviously, this guard band will lead to inefficiencies which were exaggerated in the early days since the lack of digital filtering is made it difficult to filter closely packed adjacent channels.

2.2.2 Multicarrier Communication (MC)

The concept of multicarrier (MC) communications uses a form of FDM technologies but only between a single data source and a single data receiver [23]. As multicarrier communications was introduced, it enabled an increase in the overall capacity of communications, thereby increasing the overall throughput. Referring to MC as FDM, however, is somewhat misleading since the concept of multiplexing refers to the ability to add signals together. MC is actually the concept of splitting a signal into a number of signals, modulating each of these new signals over its own frequency channel, multiplexing these different frequency channels together in an FDM manner; feeding the received signal via a receiving antenna into a demultiplexer that feeds the different frequency channels to different receivers and combining the data output of the receivers to form the received signal.

2.2.3 Orthogonal Frequency Division Multiplexing

Orthogonal Frequency Division Multiplexing (OFDM) is simply defined as a form of multi-carrier modulation where the carrier spacing is carefully selected so that each sub-carrier is orthogonal to the other sub-carriers. Orthogonality can be achieved by carefully selecting the sub-carrier frequencies. One of the ways is to select sub-carrier frequencies such that they are harmonics to each other.

2.3 Principle of ofdm transmission technology

As stated above OFDM is a multi-carrier modulation technology where every sub-carrier is orthogonal to each other. The "orthogonal" part of the OFDM name indicates that there is a precise mathematical relationship between the frequencies of the carriers in the system. It is possible to arrange the carriers in an OFDM Signal so that the sidebands of the individual carriers overlap and the signals can still be received without adjacent carriers' interference.

In order to do this the carriers must be mathematically orthogonal. Two signals are orthogonal if their dot product is zero. That is, if we take two signals multiply them together and if their integral over an interval is zero, then two signals are orthogonal in that interval. Since the carriers are all sine/cosine wave, we know that area under one period of a sine or a cosine wave is zero which is as shown below.

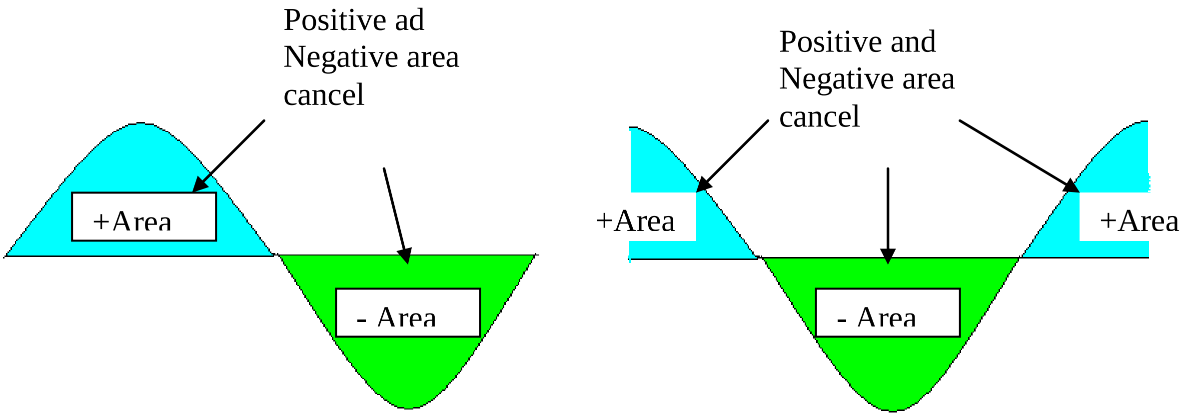

Figure. 2.1 - The area under a sine and a cosine wave over one period is always zero.

If a sine wave of frequency m is multiplied by a sinusoid (sine or cosine) of a frequency n, then the product is given by

$$f(t) = \sin m\omega t \times \sin n\omega t \qquad (2.1)$$

Where, m and n are integers. By simple trigonometric relationship, this is equal to a sum of two sinusoids of frequencies (n-m) and (n+m). Since these two components are each a sinusoid, the integral is equal to zero over one period. The integral or area under this product is given by

$$= \int_0^{2\pi} \frac{1}{2}\cos(m-n)\omega t - \int_0^{2\pi} \frac{1}{2}\cos(m+n)\omega t$$

=0-0

So when a sinusoid of frequency *n* multiplied by a sinusoid of frequency *m*, the area under the product is zero. In general for all integers *n* and *m*, sin *mx*, cos *mx*, cos *nx*, sin *nx* are all orthogonal to each other. These frequencies are called harmonics.

As the sub carriers are orthogonal, the spectrum of each carrier has a null at the center frequency of each of the other carriers in the system. This results in no interference between the carriers, allowing them to be spaced as close as theoretically possible. The orthogonality allows simultaneous transmission on a lot of sub-carriers in a tight frequency space without interference from each other as shown in Figure 2.3. So in the receiver side we can easily extract the individual sub-carriers. But in traditional FDM systems overlapping of carriers is not possible, rather a guard band is provided between each carrier to avoid inter-carrier interference which is as shown below

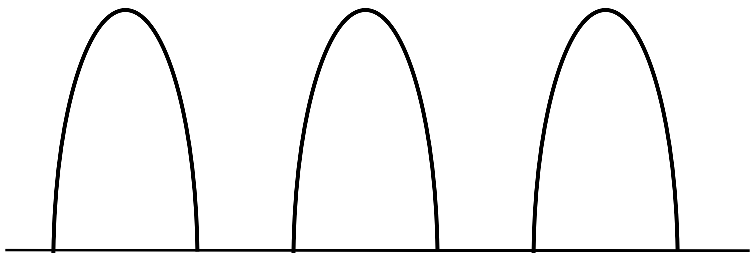

Figure. 2.2 - Spectrum of FDM

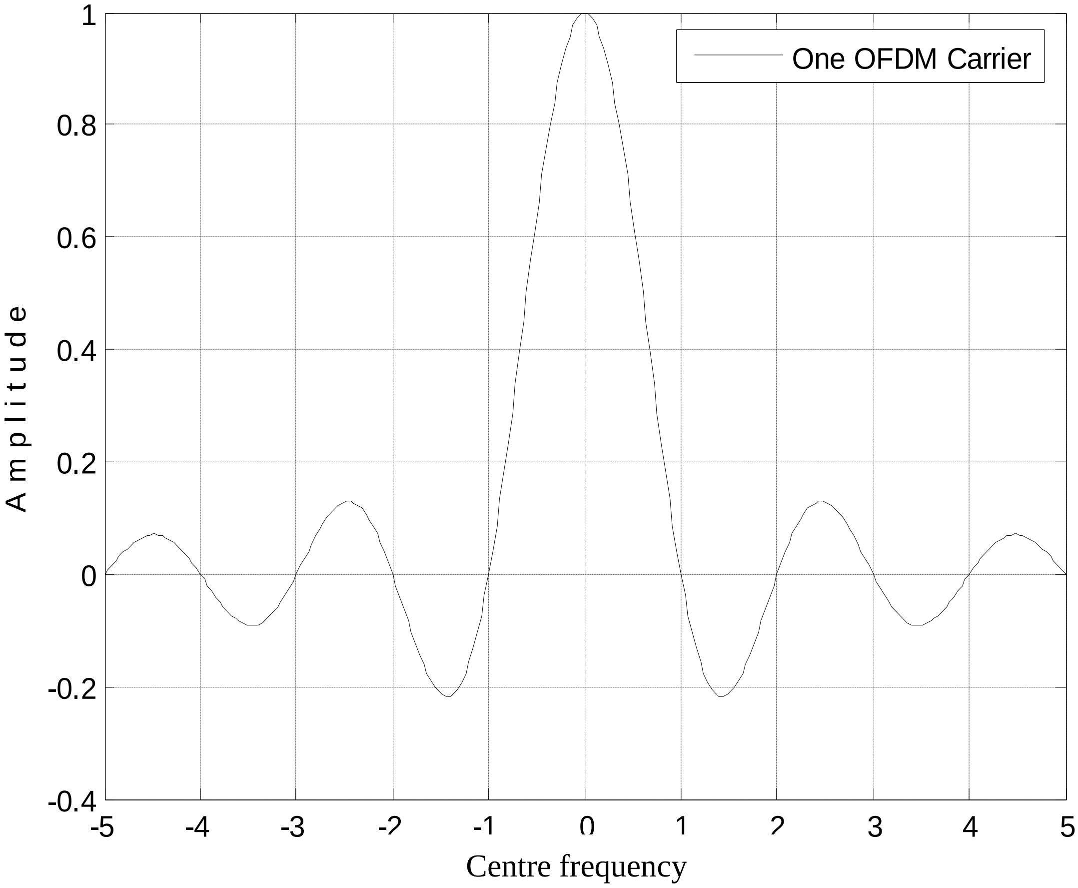

Figure. 2.3 - Single Carrier of OFDM Signal

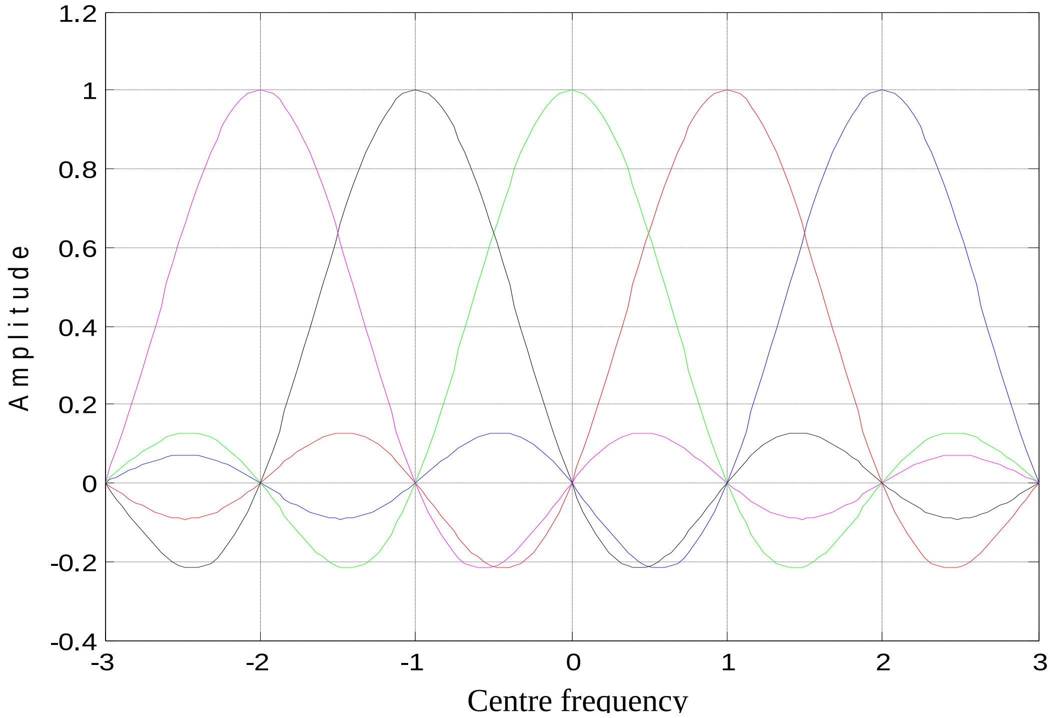

Figure. 2.4 - 5 carriers of OFDM Signal

2.4 Ofdm generation and reception

Figure 2.5 shows the block diagram of a typical OFDM transceiver. The transmitter section converts digital data to be transmitted, into a mapping of subcarrier amplitude and phase. It then transforms this spectral representation of the data into the time domain using an Inverse Discrete Fourier Transform (IDFT).

The Inverse Fast Fourier Transform (IFFT) performs the same operations as an IDFT, except that it is much more computationally efficiency, and so is used in all practical systems. In order to transmit the OFDM signal the calculated time domain signal is then mixed up to the required frequency. The receiver performs the reverse operation of the transmitter, mixing the RF signal to base band for processing, then using a Fast Fourier Transform (FFT) to analyses the signal in the frequency domain.

The amplitude and phase of the subcarriers is then picked out and converted back to digital data. The IFFT and the FFT are complementary function and the most appropriate term depends on whether the signal is being received or generated. In cases where the signal is independent of this distinction then the term FFT and IFFT is used interchangeably.

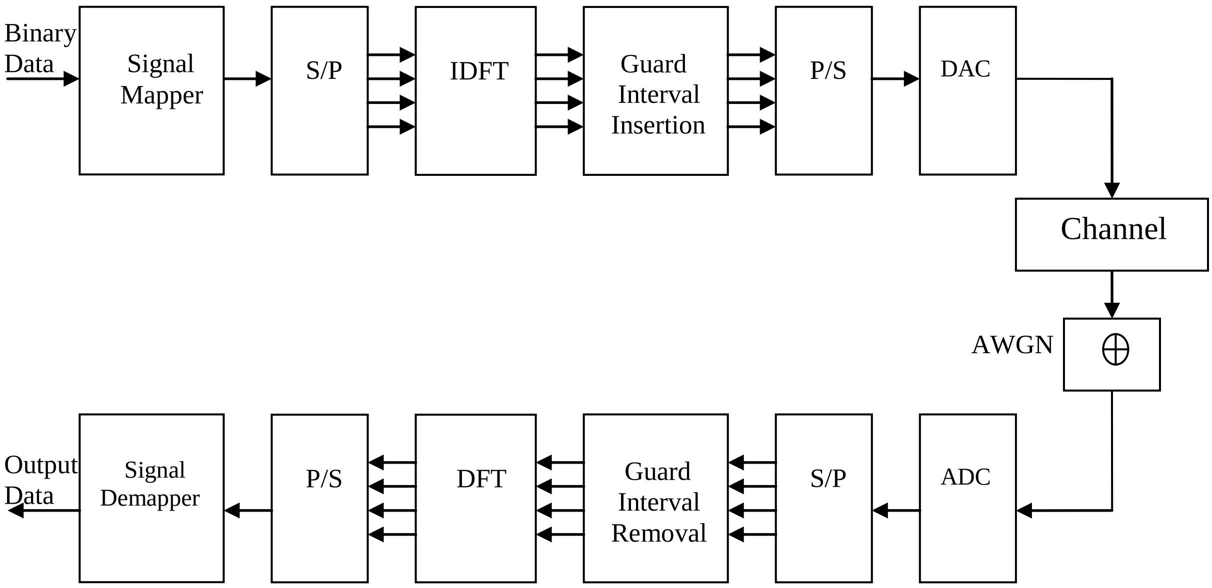

Figure. 2.5 – The basic block diagram of an OFDM system

2.4.1 Signal Mapping

A large number of modulation schemes are available allowing the number of bits transmitted per carrier per symbol to be varied. Digital data is transferred in an OFDM link by using a modulation scheme on each subcarrier. A modulation scheme is a mapping of data words to a real (In phase) and imaginary (Quadrature) constellation, also known as an IQ constellation. For example 256-QAM (Quadrature Amplitude Modulation) has 256 IQ points in the constellation constructed in a square with 16 evenly spaced columns in the real axis and 16 rows in the imaginary axis. This technique transfers 8 bits per symbol. Increasing the number of points in the constellation does not change the bandwidth of the transmission, thus using a modulation scheme with a large number of constellation points, allows for improved spectral efficiency.

For example 256-QAM has a spectral efficiency of 8 b/s/Hz, compared with only 1 b/s/Hz for BPSK. However, the greater the number of points in the modulation constellation, the harder they are to resolve at the receiver.

2.4.2 Serial to Parallel Conversion(S/P)

Data to be transmitted is typically in the form of a serial data stream. In OFDM, each symbol typically transmits 40 - 4000 bits, and so a serial to parallel conversion stage is needed to convert the input serial bit stream to the data to be transmitted in each OFDM symbol. The data allocated to each symbol depends on the modulation scheme used and the number of subcarriers. For example, for a subcarrier modulation of 16-QAM each subcarrier carries 4 bits of data, and so for a transmission using 100 subcarriers the number of bits per symbol would be 400. At the receiver the reverse process takes place, with the data from the subcarriers being converted back to the original serial data stream.

2.4.3 Frequency to Time Domain Conversion

The OFDM message is generated in the complex baseband. Each symbol is modulated onto the corresponding subcarrier using variants of phase shift keying (PSK) or different forms of Quadrature amplitude modulation (QAM). The data symbols are converted from serial to parallel before data transmission. The frequency spacing between adjacent subcarriers is $N_{\frac{\pi}{2}}$, where N is the number of subcarriers. This can be achieved by using the inverse discrete Fourier transform (IDFT), easily implemented as the inverse fast Fourier transform (IFFT) operation As a result; the OFDM symbol generated for an N-subcarrier system translates into N samples, with the ith sample being

$$x_i = \sum_{n=0}^{N-1} C_n \exp\left\{ j \frac{2\pi i n}{N} \right\}, \qquad 0 \le i \le N-1 \tag{2.2}$$

At the receiver, the OFDM message goes through the exact opposite operation in the discrete Fourier transform (DFT) to take the corrupted symbols from a time domain form into the frequency domain. In practice, the baseband OFDM receiver performs the fast Fourier transform (FFT) of the receive message to recover the information that was originally sent.

2.5 Intersymbol interference

In a multipath environment, a transmitted symbol takes different times to reach the receiver through different propagation paths. From the receiver's point of view, the channel introduces time dispersion in which the duration of the received symbol is stretched. Extending the symbol duration causes the current received symbol to overlap previous received symbols and results in intersymbol interference (ISI). In OFDM, ISI usually refers to interference of an OFDM symbol by previous OFDM symbols.

For a given system bandwidth the symbol rate for an OFDM signal is much lower than a single carrier transmission scheme. For example for a single carrier BPSK modulation, the symbol rate corresponds to the bit rate of the transmission. However for OFDM the system bandwidth is broken up into N subcarriers, resulting in a symbol rate that is N times lower than the single carrier transmission. This low symbol rate makes OFDM naturally resistant to effects of Inter-Symbol Interference (ISI) caused by multipath propagation. Multipath propagation is caused by the radio transmission signal reflecting off objects in the propagation environment, such as walls, buildings, mountains, etc. These multiple signals arrive at the receiver at different times due to the transmission distances being different. This spreads the symbol boundaries causing energy leakage between them.

2.6 Guard period

The effect of ISI on an OFDM signal can be further improved by the addition of a guard period to the start of each symbol. This guard period is a cyclic copy that extends the length of the symbol waveform. Each subcarrier, in the data section of the symbol, (i.e. the OFDM symbol with no guard period added, which is equal to the length of the IFFT size used to generate the signal) has an integer number of cycles. Because of this, placing copies of the symbol end-to-end results in a continuous signal, with no discontinuities. Thus by copying the end of a symbol and appending this to the start results in a longer symbol time.

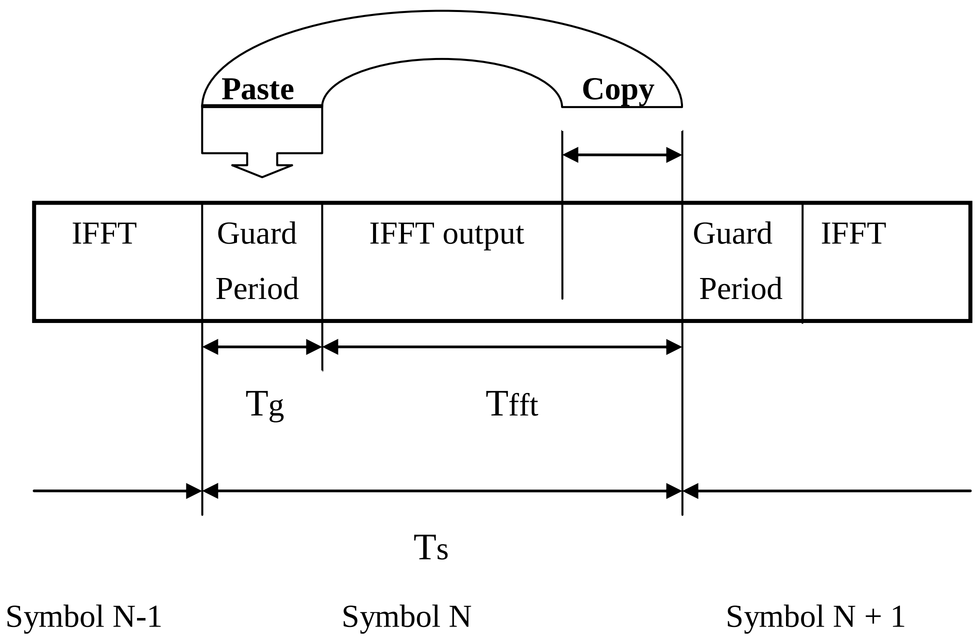

Figure. 2.6 - Guard period insertion in OFDM

Figure 2.6 shows the insertion of a guard period. The total length of the symbol is $T_S=T_G+T_{FFT}$, where T_S is the total length of the symbol in samples, T_G is the length of the guard period in samples, and T_{FFT} is the size of the IFFT used to generate the OFDM signal. In addition to protecting the OFDM from ISI, the guard period also provides protection against time-offset errors in the receiver.

2.7 Affect of AWGN on OFDM

Noise exists in all communications systems operating over an analog physical channel, such as radio. The main sources are thermal background noise, and electrical noise in the receiver amplifiers, and inter-cellular interference.

In addition to this noise can also be generated internally to the communications system as a result of Inter-Symbol Interference (ISI), Inter-Carrier Interference (ICI), and Inter- Modulation Distortion (IMD). These sources of noise decrease the Signal to Noise Ratio (SNR), ultimately limiting the spectral efficiency of the system. Noise, in all its forms, is the main detrimental effect in most radio communication systems. It is therefore important to study the effects of noise on the communications error rate and some the tradeoffs that exist between the level of noise and system spectral efficiency.

Most types of noise present in radio communication systems can be modeled accurately using additive white Gaussian noise (AWGN). This noise has a uniform spectral density (making it white), and a Gaussian distribution in amplitude (this is also referred to as a normal distribution). Thermal and electrical noise from amplification, primarily have AWGN properties, allowing them to be modeled accurately with AWGN. Also most other noise sources have AWGN properties due to the transmission being OFDM.

OFDM signals have a flat spectral density and a Gaussian amplitude distribution provided that the number of carriers is large (greater than about 20 subcarriers), because of this the inter-cellular interference from other OFDM systems have AWGN properties. For the same reason ICI, ISO, and IMD also have AWGN properties for OFDM signals.

2.8 Advantages of ofdm

Following are the advantages of OFDM

(i)Efficient use of the spectrum by allowing overlap

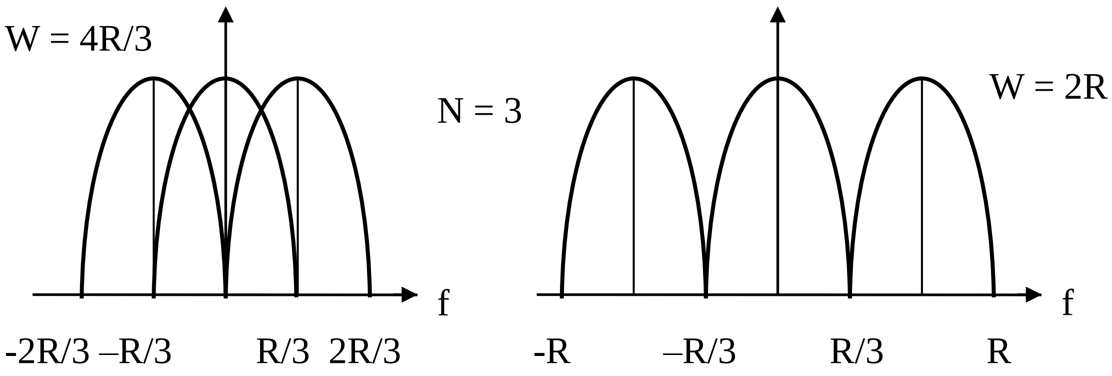

Figure. 2.7 - Spectrum Efficiency of OFDM Compared to FDM

OFDM achieves high spectral efficiency by allowing the sub-carriers to overlap in the frequency domain.

If the number of subcarriers is N and T_s is symbol duration, then total bandwidth required is

$$BW_{total} = \frac{N+1}{Ts} \tag{2.3}$$

On the other hand, the bandwidth required for serial transmission of the same data is

$$BW_{total} = \frac{2N}{Ts} \tag{2.4}$$

(ii) Robustness to Frequency Selective fading channels

In a multipath channel the reflected signals that are delayed form and get added to signal and cause either gains in the signal strength or loss (deep fade) in the signal strength. Deep fade means the signal is nearly wiped out.

A channel with deep fades occurs at selected frequencies is called a frequency selective fading channel those frequencies depends upon the environment. In a single carrier system the entire signal is lost during the fading intervals.

But as in case of OFDM the signal consists of many sub-carriers, so only few sub-carriers are affected during the fading intervals hence a very small percentage of the signal is lost which can be easily recovered.

(iii) Multipath Delay Spread Tolerance

OFDM is highly immune to multipath delay spread that causes inter-symbol interference in wireless channels. Since the symbol duration is made larger (by converting a high data rate signal into ‘N’ low rate signals), the effect of delay spread is reduced by the same factor. Also by introducing the concepts of guard time and cyclic extension, the effects of intersymbol interference (ISI) are removed completely.

2.9 Applications of OFDM

OFDM forms the basis for the Digital Audio Broadcasting (DAB).

OFDM forms the basis for the Digital Video Broadcasting (DVB) and HDTV.

It is used for wideband data communications over mobile radio channels such as High-bit-rate Digital Subscriber Lines (HDSL at 1.6Mbps), Asymmetric Digital Subscriber Lines (ADSL up to 6Mbps) and Very-high-speed Digital Subscriber Lines (VDSL at 100 Mbps).

It is used in HiperLAN/2, which is a European Wireless LAN standard.

The IEEE 802.11 working group published IEEE 802.11a, which outlines the use of OFDM in the 5GHz band.

OFDM is under consideration for use in 4G Wireless systems

2.10 Disadvantages of OFDM

Two major drawback of OFDM systems are

(i) ICI (Inter carrier Interference) which is due to frequency offset.

Considering orthogonality of the sub-carriers, are able to extract the symbols at the receiver do not interfere with each other.

Orthogonality is preserved as long as sub-carriers are harmonics to each other. But if at the receiver end there is a change of frequency of the sub-carriers due to any reason then the orthogonality among them is lost & ICI occurs. As a result the signal degrades heavily.

This change in frequency is called frequency offset. There are two main reasons for frequency offset.

(a) Frequency mismatch between transmitter & receiver

(b) Doppler effect

(ii) PAPR (Peak-to-Average Power Ratio)

It is because the time domain OFDM signal is a summation of several orthogonal sub-carriers, so OFDM signal has high variation in its envelope.

CHAPTER 3

MULTIPLE INPUT MULTIPLE OUTPUT-OFDM

3.1 Why MIMO-OFDM?

MIMO can be used with any modulation scheme or access technique. Today, most digital radio systems use Time Division Multiple Access (TDMA), Code Division Multiple Access (CDMA) and Orthogonal Frequency Division Multiplexing (OFDM).

Time division systems transmit bits over a narrowband channel, using time slots to segregate bits for different users or channels. Code division systems transmit bits over a wideband (spread spectrum) channel, using codes to segregate bits for different users or purposes. OFDM is also a wideband system, but unlike CDMA which spreads the signal continuously over the entire channel, OFDM employs multiple, discrete, lower data rate sub channels.

MIMO can be used with any modulation or access technique. However, research shows that implementation is much simpler—particularly at high data rates—for MIMO-OFDM. Specifically, MIMO-OFDM signals can be processed using relatively straightforward matrix algebra.

3.2 What is the Best Way to Deliver Wireless LAN Innovations?

One of today's biggest high-tech controversies is how best to encourage and deliver technology innovations. Everyone agrees that standards play an important role in ensuring interoperability and cost reduction (through volume production of key components). Not everyone agrees, however, about what makes some standards more successful than others. In particular, there is difference of opinion regarding whether standards drive innovation or innovation drives standards.

Each year, thousands of standards are published. Some standards are extremely successful; many achieve limited success. But the majority is simply overtaken by events. The Global Systems for Mobile Communications (GSM) is an example of an extremely successful standard. Internet Protocol Version 6 (IPv6) is almost universally recognized as the Internet's next generation protocol, but it has made little headway in replacing IPv4. The Internet was built on TCP/IP, a protocol developed by two engineers, instead of the highly-touted Open Systems Interconnection (OSI) protocol developed by the International Standards Organization (ISO).

Successful standards often codify market-proven innovations. Ethernet, like Token Ring, began life as a proprietary local area networking (LAN) standard. Vendors have learned over time that it’s better to control 20% of a very large market than 100% of a small market with a doubtful future. That’s why most vendors are anxious to offer a proprietary innovation as the foundation for an open standard. To wit, most successful standards start with a technology that has gained market traction and open it to wider participation. Standards for proven markets deserve and receive more attention and urgency. Participants are more amenable to compromise, recognizing that neither vendors nor users benefit from lengthy delays. Some commentators and vendors warn that bringing innovations to market before relevant standards have been published hinders market development.

However, this claim is not supported by the facts. Today, there are many successful standards that started as proprietary innovations with clear end-user benefits. But there are few if any examples of markets that failed to develop simply because such innovations were introduced maturely. Another common complaint is that proprietary and “pre-standard” products are harmful to customers. Specifically, it is asserted that these customers will wake up one day to learn that the products they invested in are not compatible with the published standard. However, this argument overlooks several important points. Many customers buy proprietary or pre-standard products because they have specific, urgent requirements addressed by such solutions. Standards usually take a minimum of 1-2 years to develop, and products conforming to those standards usually require another year before they are commercially available; many customers routinely replace computer and communications equipment every few years to take advantage of further developments.

More importantly, vendors have become more sophisticated about introducing innovative technology. For example, wireless LAN products incorporating MIMOOFDM technology (often called “pre-standard, MIMO-enhanced Wi-Fi” because MIMOOFDM is central to the forthcoming 802.11n standard) are fully compatible and interoperable with existing 802.11b, 802.11g, and 802.11a devices on the same network at the same time.

Plus, it’s not unusual for vendors that sell pre-standard enhancements to introduce products incorporating both standard and pre-standard modes once the standard has been finalized, providing their customers a bridge between the two environments. For example, in the late 1980s dial-up modem maker US Robotics offered a proprietary high-speed mode; once the V.32 high-speed standard was published, US Robotics introduced what it dubbed its “dual standard” product. The upshot is that innovation drives standards—not the other way around. Proprietary and pre-standard enhancements play a pioneering and positive role. The vision of hordes of customers left stranded by vendors of proprietary or pre-standard enhancements is a myth. The more successful a pre-standard enhancement is, the more incentive for the vendor to continue supporting it by developing products that include both pre-standard and standard modes.

3.3 How MIMO Benefits Products Based on Existing Standards

The fact that products incorporating pre-standard MIMO-enhanced Wi-Fi look the same on the outside as products with smart antennas hints at another benefit. The pre-standard MIMO products have the antennas and other circuitry needed to provide smart antenna functionality when operating in a mode based on existing standards. In fact, this smart antenna functionality boosts performance even when present on just one end of a link. Thus, pre-standard MIMO products are not only fully compatible with existing Wi-Fi standards; they also enhance the performance of those standards.

This demonstrates in yet another way that pre-standard MIMO enhancements can be designed to complement rather than thwart standards. Thus, while customers may purchase pre-standard MIMO-enhanced Wi-Fi products to take advantage of MIMOOFDM's superior performance, some may discover that merely adding these devices to a standards-based network does the job for them

3.4 MIMO-OFDM Enables Wireless LAN Applications and Markets

Wireless LANs scored their first major success in vertical industrial applications—primarily warehouse and retail floor inventory management. The market exploded as wireless LANs were embraced for PC networking and sharing broadband access in small businesses and homes. This success has positioned wireless LANs to drive the development of three new Markets with huge growth potential: home entertainment networking, cordless Voice over IP (VoIP), and a variety of machine-to-machine (M2M) applications. Currently, tens of millions of wireless LAN nodes are shipped annually. Home entertainment applications present the opportunity to sell hundreds of millions of nodes per year—one for every television, stereo system, DVD player, remote screen, remote speaker, and portable record/playback device shipped.

Cordless VoIP represents an even greater prospect. Assuming a significant fraction of private and public wireless LANs are modified to handle VoIP traffic, a market for integrating Wi-Fi with mobile phones is operated to emerge. There are currently 1.5 billion Mobile phone subscribers, with more than 500 million handsets sold annually. And there is more to cordless VoIP than first meets the eye; for example, cordless VoIP could enable strategic alliances between mobile phone and cable network operators. Major Cable operators are entering local phone markets, creating opportunities for mobile carriers to offer handsets that serve as cordless phones in the home a single phone for all of users' telephone service needs.

The potential number of wireless LAN nodes needed for machine-to-machine applications is mind boggling. For example, the average home security system could easily use a dozen nodes for door and window sensors, motion detectors, and video cams. Other major wireless M2M applications include telematics, asset monitoring, mobile commerce, and healthcare and real-time enterprise communications.

3.5 MIMO OFDM TRANCEIVER

To meet the ever growing demand for higher data rates in wireless communication systems, multiple transmit and receive antennas can be employed to make use of the spatial dimension by transmitting data in parallel streams using space time block coding. Such spatial multiplexing Multiple-Input Multiple-Output (MIMO) systems have been shown to obtain significantly higher data rates than Single-Input Single-Output (SISO) systems. This increase in data rate can be achieved without the need of additional bandwidth or transmit power, provided that sufficient multipath diversity is present [26, 27].

Orthogonal Frequency Division Multiplexing (OFDM) transforms a frequency selective (time dispersive) channel into parallel narrow band at fading channels, which reduces the equalization task to a simple scalar multiplication in each subcarrier [28]. In a spatial multiplexing system, equalization consists of separating the streams as well as combating the time dispersive nature of the MIMO channel, which is referred to as space-time equalization. Thus, the simple equalization of OFDM is an attractive feature in the context of MIMO systems, since it leads to low complexity space-time equalization. MIMO OFDM is also being considered for throughput enhancement in the emerging IEEE 802.11n wireless LAN standard.

To acquire the Channel State Information (CSI) at the receiver, training data or pilot tones known to the receiver are transmitted in traditional systems. However, this training overhead can amount to a considerable portion of the overall bandwidth. To save the valuable bandwidth, blind equalization can be employed.

Blind methods avoid the use of training by exploiting the statistics of the source streams and the MIMO channel to recover the data [30].Proposed approaches for blind detection in MIMO systems include: the constant modulus algorithm (CMA) [31, 32, 33], subspace methods [34, 35], CMA is based on the gradient descent Method, which may suffer from slow convergence and misconvergence at low (SNR) levels. Subspace methods employ

SOS, which may demonstrate poor performance with ill-conditioned MIMO channel matrices and sensitivity to Gaussian noise.

3.6 Signal Model

OFDM transmits frames of IDFT data. At the transmitter a cyclic prefix (CP) consisting of a copy of the last LCP symbols is added to each frame during the guard interval and removed at the receiver before transforming back to the frequency domain. The CP avoids inter frame interference and makes the channel convolution matrix circulant when $L_{CP} > (L_c - 1)$, where L_c is the channel impulse response length.

The Discrete Fourier Transform (DFT) property of diagonal sings any circulant matrix is used in OFDM, so that the channel electively only introduces a scalar multiplication in each sub carrier. Thus, OFDM avoids intersymbol interference (ISI). This transformation of a frequency selective channel into parallel at fading channels and the low transceiver complexity are useful properties for the application in MIMO systems. In a spatial multiplexing MIMO OFDM system N_t transmit and N_r receive antennas are employed and each transmit antenna emits data at the same time and on the same frequency band.

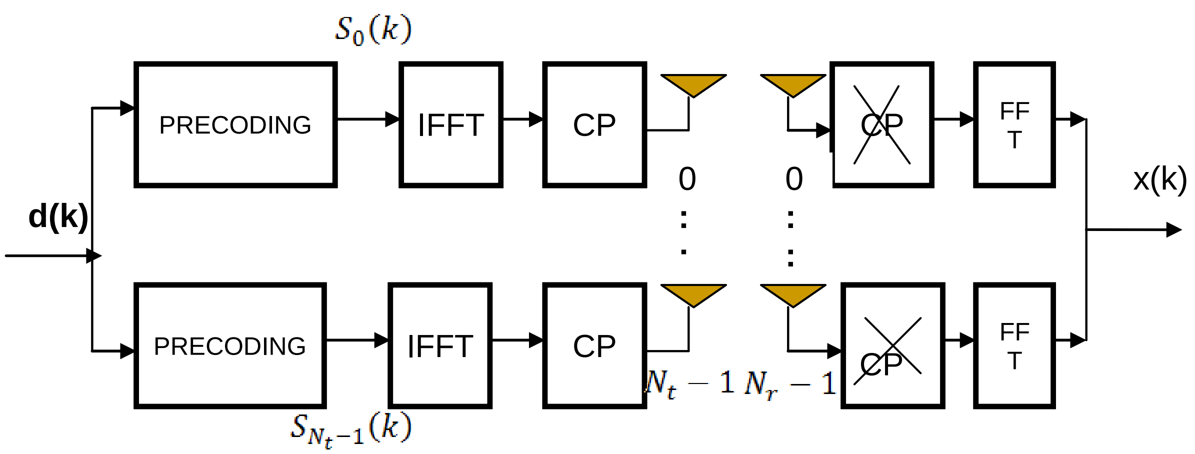

Figure 3.1 Signal model in MULTIPLE INPUT MULTIPLE OUTPUT-OFDM

Fig. 3.1 shows the overall MIMO OFDM system. At the transmitter each stream is rest preceded and then transmitted using OFDM. At the receiver side, OFDM demodulation is followed by space-time equalization, which is performed by the proposed ICA-MMSE or ICAMMSE+ SIC methods and includes decoding of the streams. The efficient Fast Fourier Transform (FFT) is used to implement the DFT

The frequency domain signal model is

$$s(k,i) = \left[s_0(k+iN), s_1(k+iN), \cdots\cdots\cdots, s_{N_t-1}(k+iN) \right] \tag{3.1}$$

'k' is obtained demultiplexing the source signal

'N' is the total no of sub carriers

After removal of the CP and application of the DFT at the receiver, the ith received OFDM frame on sub carrier 'k' is

$$x(k,i) = H(x)s(x) + n(k,i) \tag{3.2}$$

Where $x(k,i) = \left[x_0(k+iN), x_1(k+iN), \cdots\cdots\cdots, x_{Nr-1}(k+iN) \right]$

here n(k,i) is AWGN has zero mean and variance and the element $H_{r,t}(x)$ on row 'r' and column 't' of H(x) is a scalar channel between transmit antenna 't' and receive antenna 'r'.The element $H_{r,t}(x)$ on row r and column t of H (k) is the scalar channel gain between transmits antenna t and receives antenna r.

The element of H (k) are assumes to be i.i.d. complex random variables with Rayleigh distributed amplitude and uniformly distributed phase and remain constant for the duration of a block consisting of Ns OFDM frames, which corresponds to block fading in a multipath environment with no direct line of sight [43]. The complex valued (AWGN) vector $n(k,i)$ has zero mean and variance, which is twice the variance of the real or imaginary part.

CHAPTER 4

ICA Based Blind MIMO-OFDM

4.1Approaches for blind detection in MIMO systems

Blind detection in MIMO systems include: the constant modulus algorithm (CMA) [31, 32, 33], subspace methods [34, 35], other methods using second order statistics (SOS) [37, 38] and Independent Component Analysis (ICA) based methods [33, 38, 39]. CMA is based on the gradient descent method, which may suffer from slow convergence and misconvergence at low (SNR) levels. Subspace methods employ SOS, which may demonstrate poor performance with ill-conditioned MIMO channel matrices and sensitivity to Gaussian noise. ICA relies on higher order statistics (HOS) to recover the transmitted streams from the received mixture. Blind separation of the mixture is achieved based on the assumption of mutual statistical independence of the source streams. A useful feature of ICA is that for the equivariant class of ICA methods, the MIMO channel estimation accuracy is independent of the channel [40]. Additionally, the use of HOS makes ICA methods less sensitive to Gaussian noise than Subspace methods

ICA for MIMO systems has mainly assumed single carrier systems and/or at fading channels. In [38], ICA was employed in one subcarrier of OFDM to combat frequency selective fading channels, and a Minimum Mean Squared Error (MMSE) approach was used to recover the remaining subcarriers. However, this method results in severe error propagations over subcarriers as the subcarrier being unmixed uses the stream estimates from the previous subcarrier as a reference. The receiver proposed in [39] obtains better performance by using ICA in every subcarrier. However, since a large number of subcarriers may be used in an OFDM system and additionally a reordering step is required in each sub carrier to overcome the scaling and order indeterminacies of ICA, this receiver has a high complexity.

It is proposed two ICA based blind MIMO OFDM receivers. The first is an extension of [38], which uses ICA in one sub carrier, combined with an MMSE approach to unmixed the remaining subcarrier. However, due to a novel non-redundant encoding, this ICA-MMSE receiver provides significant performance enhancement over [38] because it does not suffer from the error propagations over subcarriers while maintaining the low computational complexity.

To further improve the performance, the second proposed receiver combines ICA with Successive Interference Cancellation (SIC) [41], which is termed ICA-MMSE+SIC.

Compared to [39], where ICA is employed in every subcarrier, the proposed receivers have a much lower computational complexity. The lower complexity is obtained because ICA is used in only one subcarrier and no reordering is required to overcome the order and scaling indeterminacies of ICA. It shows that the proposed ICA-MMSE receiver obtains similar performance to [39] and significantly outperforms the method in [38]. Besides, ICA-MMSE+SIC can obtain performance close to the case with perfect CSI at the receiver when the number of receive antennas is larger than the number of transmit antennas.

4.2 Signal model in ICA Based Blind MIMO-FDM

In a spatial multiplexing MIMO OFDM system N_t transmit and N_r receive antennas are employed and each transmit antenna emits data at the same time on the same frequency band. Fig. 4.1 shows the overall MIMO OFDM system. At the transmitter each stream is rest preceded and then transmitted using OFDM. At the receiver side, OFDM demodulation is followed by space-time equalization, which is performed ICA-MMSE or ICAMMSE+ SIC methods and includes decoding of the streams. The efficient (FFT) is used to implement the DFT

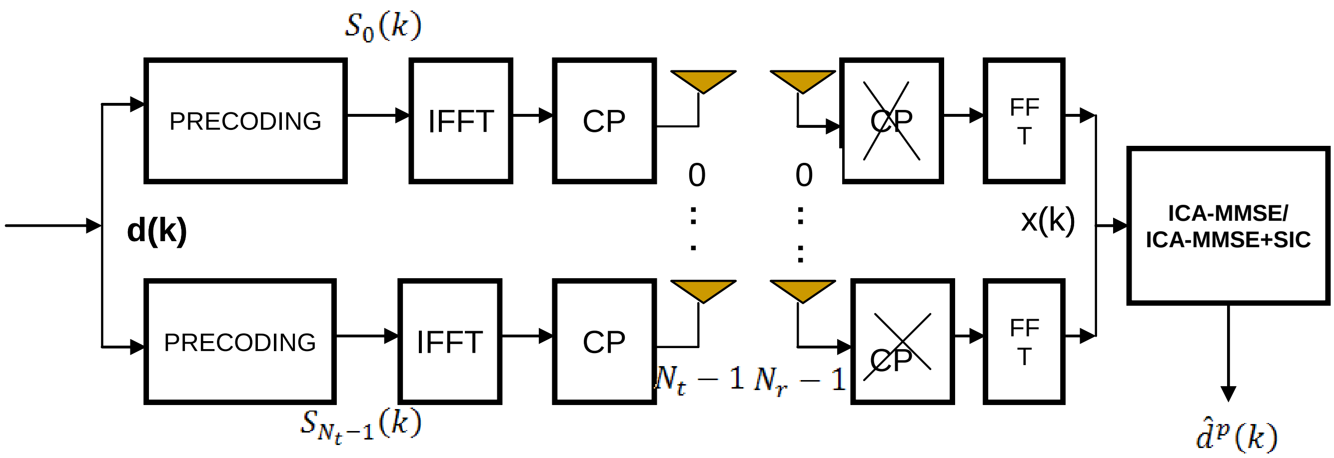

Figure 4.1 Signal model in ICA Based Blind Multiple-Input Multiple-Output OFDM

The frequency domain signal model is represented as

$$s(k,i) = \lfloor s_0(k+iN), s_1(k+iN), \cdots\cdots\cdots, s_{N_{t-1}}(k+iN) \rfloor \tag{4.1}$$

'k' is obtained demultiplexing the source signal and 'N' is the total no of carriers

After removal of the CP and application of the DFT at the receiver, the ith received OFDM frame on sub carrier 'k' is

$$x(k,i) = H(x)s(x) + n(k,i) \tag{4.2}$$

Where $x(k,i) = [x_0(k+iN), x_1(k+iN), \cdots\cdots\cdots, x_{Nr-1}(k+iN)]$

here n(k,i) is AWGN has zero mean and variance and the element $H_{r,t}(x)$ on row 'r' and column 't' of H(x) is a scalar channel between transmit antenna 't' and receive antenna 'r'.The element $H_{r,t}(x)$ on row r and column t of H (k) is the scalar channel gain between transmits antenna t and receives antenna r. The elements of H (k) is assumed to be i.i.d. complex random variables with Rayleigh distributed amplitude and uniformly distributed phase and remain constant for the duration of a block consisting of Ns OFDM frames, which corresponds to block fading in a multipath environment with no direct line of sight [43]. The complex valued (AWGN) vector n (k, i) has zero mean and variance, which is twice the variance of the real or imaginary part.

The MIMO OFDM signal model corresponds to the ICA model of instantaneous linear mixtures for each subcarrier k [38]. This allows the use of ICA for MIMO OFDM space-time equalization on a per subcarrier basis. ICA could be applied to estimate the transmitted streams $\hat{S}(k,i)$ and MIMO channels $\hat{H}(k,i)$ blindly by applying ICA in each subcarrier to the received mixture x (k, i). However, since ICA suffers from permutation and scaling ambiguities [40], the rows of $\hat{s}\ (k,i)$ and columns of $\hat{H}\ (k)$ could have a different order and phase shift in each subcarrier if ICA were applied per subcarrier.

To avoid different permutations and scaling across subcarriers and to lower the computational complexity, ICA is applied in only one subcarrier in [38], while the remaining subcarriers are unmixed using an MMSE approach with the stream estimates in the previous subcarrier as a reference for unmixing the current subcarrier.

This guarantees the same order and scaling of the data and channel estimates in all subcarriers but can lead to severe error propagation across subcarriers. We will use the same philosophy for our proposed receivers while using a modified correlation structure which does not lead to such error propagation across subcarriers. Precoding similar to [26] is used

$$s_t(k+iN) = \frac{1}{\sqrt{1+a^2}}\left[d_t(k+iN) + a d_t(k_r + iN)\right] \tag{4.3}$$

Correlation between data in subcarrier k_r and data in the remaining (N-1) subcarriers? Here $d_t(k+iN)$ is the unit variance source stream of transmit antenna t drawn from the Binary Phase Shift Keying (BPSK) alphabet {-1,+1}with the source streams acollected from all transmit antennas given by

$$d(k,i) = \left[d_0(k+iN), d_1(k+iN), \cdots\cdots d_{N_t-1}(k+iN)\right]^T \tag{4.4}$$

The constant a is a real valued scalar with a<1 and kr is the reference subcarrier. Note that any subcarrier can be used as the reference subcarrier. Data in every subcarrier except the reference subcarrier $k \in \{0,1,\cdots\cdots,N-1\} \setminus k_r$ is precoded. The precoding operation does not increase the transmit power or introduces any redundancy. Additionally, higher order constellations such as QPSK also be used.

4.3 ICA-MMSE Receiver

In this section the ICA-MMSE receiver which uses ICA only in one reference subcarrier to estimate the transmitted streams blindly. The streams in the remaining subcarriers are subsequently recovered using an MMSE method. The use of ICA in only one subcarrier coupled with an MMSE approach for the remaining subcarrier was proposed in [38]. However, our receiver does not suffer from error propagation across subcarriers as we use the same reference subcarrier for unmixing all the remaining subcarriers with the MMSE method. In this paper we use the JADE [44] method, however, any other ICA method which can handle complex valued channels and signals may be used instead, eg. The FastICA extension to complex valued signals in [35] or the closed-form estimators in [46].

The main concepts of ICA [40] with a focus on our application. Source separation is often performed in two steps. First, a whitening matrix $W(k_r)$ is sought to obtain spatially uncorrelated signals $z(k_r, i) = W(k_r)x(k_r, i)$

$$\underset{i}{E}\left\{z(k_r, i)z(k_r, i)^H\right\} = I_{N_t} \tag{4.5}$$

Where $\underset{i}{E}\left\{z(k_r, i)\right\}$denotes the expectation with respect to is the Hermitian transpose and N_t is an identity matrix of size $N_t \times N_t$. The whitening stage also performs a reduction of the dimension when the system uses more receive than transmit antennas ($N_r > N_t$).

The second step obtains an orthogonal matrix $V(k_r)$ such that the separating matrix $G(k_r) \cong V^H(k_r)W(k_r)$ recovers the transmitted streams

$$\hat{s}(k_r, i) = G(k_r)x(k_r, i) \tag{4.6}$$

The separating matrix can be found by minimizing the mutual information between the entries of the stream estimates $\hat{s}(k_r, i)$. In ICA, the stream estimates are related to the true streams under the assumption of no noise and perfect separation by

$$\hat{s}(k_r, i) = PDs(k, i) \tag{4.7}$$

Where the permutation matrix P accounts for the order indeterminacy and the non-singular diagonal matrix D for the scaling indeterminacy of ICA. The separating matrices G (k) for the remaining subcarriers $k \in \{0,1,\cdots\cdots, N-1\} \setminus k_r$ are obtained with the MMSE as approached in [38], which exploits the correlation structure introduced in

$$G(k) = \beta R_{xs}^H(k, k_r)R_{xx}^{-1}(k) \tag{4.8}$$

$$R_{xs}(k, k_r) = \underset{i}{E}\left\{x(k, i)\right\}\hat{s}(k_r, i)^H \text{ And}$$
$$R_{xx}(k) = \underset{i}{E}\left\{x(k, i)\right\}x(k, i)^H$$

The MMSE method obtains the same order and scaling of the stream estimates $\hat{S}(k, i)$ as in the reference subcarrier k_r when the scalar constant β is set to [38]

$$\beta = \frac{\sqrt{1+a^2}}{a}$$

Where a is the precoding constant.

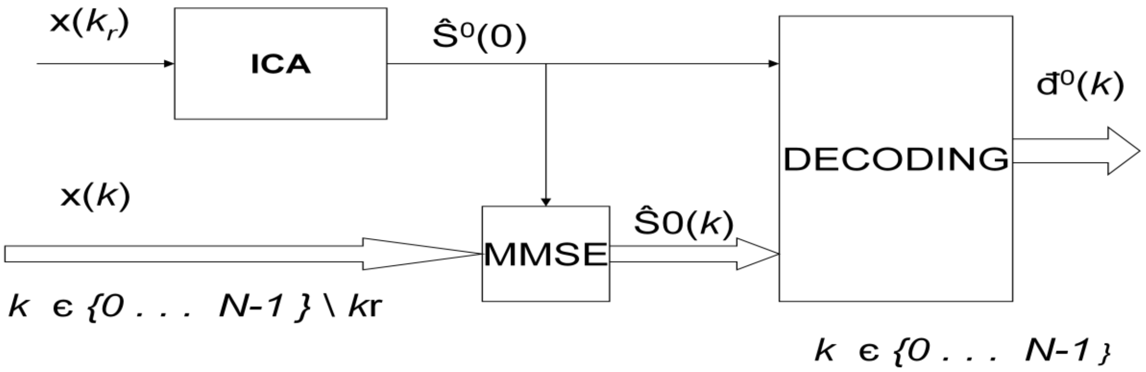

Figure 4. 2. ICA-MMSE receiver with ICA used in the reference subcarrier kr,

Fig. 4.2 shows the proposed ICA-MMSE receiver, with ICA in one subcarrier coupled with the MMSE method for the remaining subcarriers to obtain the encoded soft stream estimates $\hat{s}^p(k)$, followed by a decoding stage which outputs the hard estimated and encoded stream estimates $\hat{d}^p(k)$. Here P is the number of iterations of SIC, with P = 0 denoting the ICA-MMSE stage described in this Section. The time index i has been dropped to simplify the notation.

4.4 decoding

After separation of the mixtures, the soft stream estimates $\hat{s}^0(k,i)$ are decoded and hard estimated to obtain the final estimates $\hat{d}^0(k,i)$ .The decoded soft stream estimates are

$$\tilde{d}_t^0(k+iN) = \sqrt{1+a^2}\,\hat{s}_t^0(k+iN) - a\hat{s}_t^0(k_r+iN) \tag{4.9}$$

It may be note that the data in the reference subcarrier kr is not precoded, i.e.

$$\tilde{d}_t^0(k_r+iN) = \hat{s}_t^0(k_r+iN).$$

The hard estimates are

$$\hat{d}^0(k,i) = \left[\hat{d}_0^0(k+iN), \hat{d}_1^0(k+iN), \cdots\cdots \hat{d}_{N_t-1}^0(k+iN)\right]^T \qquad (4.10)$$

$$\hat{d}_t^0(k+iN) = Q\left[\tilde{d}_t^0(k+iN)\alpha_t\right]$$

With the hard estimation function for BPSK data, which maps the soft estimates to the BPSK alphabet {-1, +1}, before hard estimation, the data has to be derogated by $\alpha_t = \underset{k,i}{E}\left\{\left[\tilde{d}_t^0(k+iN)\right]^2\right\}^{-\frac{1}{2}}$ with the expectation across all subcarriers k and frame indices i. It may be note that for decoding, the reference subcarrier k_r has to be recovered previously. The hard estimates $\hat{d}^0(k,i)$ recover the true streams up to a stream and quadrant ambiguity

4.5 ICA-MMSE+SIC Receiver

The second receiver termed ICA-MMSE+SIC, uses SIC as a second stage to the ICA-MMSE receiver. Compared to linear equalizers such as zero forcing (ZF) or MMSE equalizers, non-linear counterparts using SIC [41, 47, 48] can obtain improved performance in MIMO systems [49]. As with the ICA-MMSE receiver, we will apply SIC on a per subcarrier basis. In a SIC equalizer, the streams are extracted one at a time, ordered from lowest to highest post-detection Mean Squared Error (MSE). After extraction, decoding and hard estimation of each stream, its contribution is removed from the received mixture. It was shown in [41] that this ordering minimizes the MSE of the stream with the largest MSE value, which predominates the system error performance. Furthermore, SIC allows making use of the discrete nature of the streams.

The SIC scheme, summarized below:

1. Obtain tentative stream estimates with ICA-MMSE.
2. Estimate the MIMO channels using these tentative stream estimates.
3. Refine the stream estimates with SIC. Steps 2 and 3 can be repeated to improve the performance further.

The ICA-MMSE+SIC receiver is depicted in Fig. 4. 3, where the time index i has been dropped. The tentative stream estimates from the ICAMMSE receiver in Section 3, denoted by iteration p = 0, are used to estimate the MIMO channel matrices H (k), which are subsequently employed for SIC

in iterations p = 1,………….,P.

The Least Squares (LS) MIMO channel estimate is

$$\hat{H}^{p}(k) = X(k)\left[\bar{S}^{p-1}(k)\right]^{+} \qquad (4.11)$$

$X(k) = \left[x(k,0), x(k,1), \ldots\ldots, x(k, N_s - 1)\right]$ is the blocked received signal

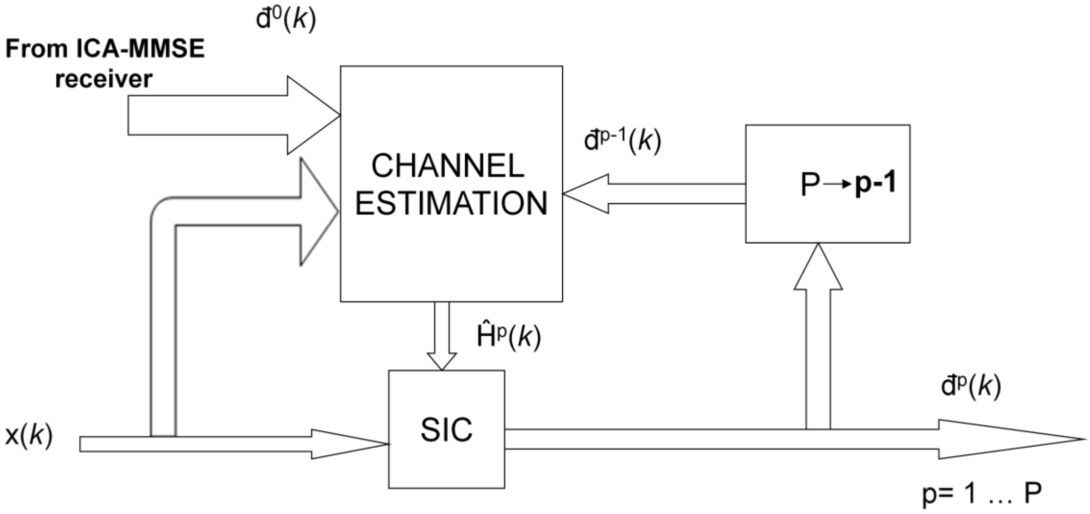

Figure.4 3. ICA-MMSE+SIC receiver using SIC as a second stage to ICA-MMSE used for initialization of the channel estimation

Finally, the streams are decoded to obtain the soft estimates $\hat{d}_t^{p}(k+iN)$

The hard estimated decoded streams are

$$\hat{d}_t^{p}(k+iN) = Q\left[\tilde{d}_t^{p}(k+iN)\right] \qquad (4.12)$$

Before the extraction of each stream, the interference of the previously detected stream is cancelled from the received mixture

$$x(k,i) = x(k,i) - \hat{h}_t^{p}(k)\bar{s}_t^{p}(k+iN) \qquad (4.13)$$

Where $\bar{s}_t^{p}(k+iN)$ is the re-encoded stream

CHAPTER 5

FPGA IMPLEMENTATION OF ICA

5.1 INTRODUCTION

Independent component analysis (ICA) [51], [52] recovers independent source signals from their mixtures by finding a linear transformation that can maximize the mutual independence of mixtures. The ICA algorithm can be applied to signal process applications such as electroencephalograms (EEG), magneto encephalograms (MEG) [53], and the famous cocktail-party problem. Some applications need real-time signal processing such as speech signal enhancement, noise canceling, and electrocardiogram (ECG) signal analysis. However, related papers published in recent years almost are focus on off-line signal analysis. Therefore, this study examines ICA implementation on a FPGA chip for real-time signal process applications.

Field programmable gate array (FPGA) technology can implement the digital signal processing algorithm and quickly verify the result in hardware. Currently, most FPGAs have on-chip hardware multipliers and memory blocks, so FPGAs fit in with the implementation of the ICA algorithm that require high volumes of mathematic operations. Very high speed integrated circuit hardware description language (VHDL) is used to design the hardware.

Many scientific problems require floating-point (FP) arithmetic [54] with high accuracy in their calculations and a large dynamic range of numbers is necessary for such signal processing techniques. Nevertheless, FP is difficult to implement on the FPGA, because the FP algorithm is too complex and the area (logic-elements) of FPGA lead to excessive consumption when implemented. Shirazi et. al [55] developed the 18-bit FP format to minimize the size of implementation, although its accuracy cannot satisfy the FastICA algorithm that requires high calculation throughput. Therefore, simplified 32-bit FP implementations.

There are no publications on the implementation of the FastICA algorithm. Charoensak and Sattar [56] focus on the implementation of BSS using Torkkola's network with 8 kHz sample rate and 64.4 MHz system clock. We have presented the hardware FastICA using a lower system clock but a higher sample rate than Charoensak and Sattar.

5.2 BACKGROUND OF THE FASTICA ALGORITHM

5.2.1 Preprocess of FastICA

Let the measured mixing signals V1 and V2 be expressed as

$$v_1 = (v_{11}, v_{12}.........v_{1(m-1)}, v_{1(m)}) \tag{5.1}$$

$$v_2 = (v_{21}, v_{22}.........v_{2(m-1)}, v_{2(m)}) \tag{5.2}$$

Where *m* is the length of the mixing signal. Assume the mixing signal **V** is defined as

$$\text{V} = \text{AS} \tag{5.3}$$

Where $V = (v_1, v_2)$**A** is the unknown full rank mixing matrix and $S = (s_1, s_2)$ is the unknown source signal that is statistically independent. The goal of ICA is to recover the source signal **S** by estimating the mixing matrix **A** as illustrated in the Figure 1. In order to reduce the complexity of the ICA method, it is necessary to preprocess the measured signal **V** before ICA. The first step of preprocess is called centering, and its method is to calculate the sample mean μ_i from measured signal v_i and then subtract μ_i from v_i

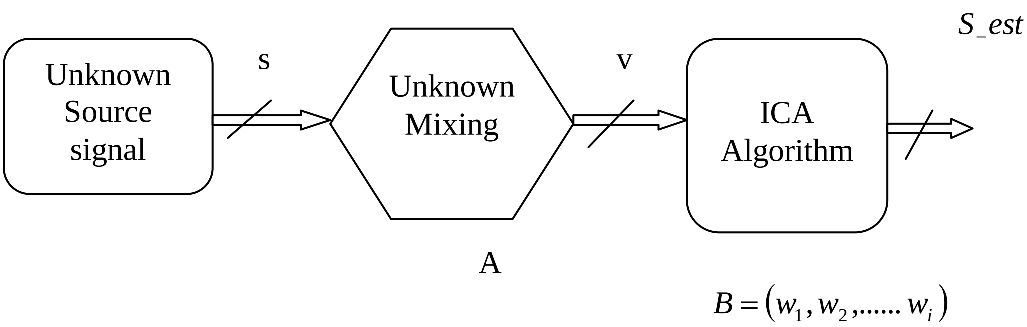

Figure.5.1.ICA signal process

Sample means of measured signals are defined as

$$\mu_1 = E\{v_1\} \\ \mu_2 = E\{v_2\} \tag{5.4}$$

After centering, **V** becomes a zero mean matrix.

The second step of preprocessing is called whitening, which can be described as

$$X = PV \tag{5.5}$$

Where **P** is a whitening matrix and **X** is the whitened data. After whitening, elements in the matrix $X = (x_1, x_2)$ are uncorrelated and the covariance is equal to the identity matrix, that is $C_x = E\{XX^T\} = I$ The whitening matrix P is given by $D^{\frac{-1}{2} \times E^T}$,where $D = diag(d_1, d_2)$ is a diagonal matrix of the eigenvalues that were calculated from the covariance matrix $C_v = E\{VV^T\}$,and $E = (e_1, e_2)$ is the orthogonal matrix of C_v's eigenvectors.

5.2.2 Algorithm of FastICA

After preprocessing, FastICA will estimate the source signal $S\_est = (s\_est_1, s\_est_2)$ from **X** by estimating the matrix **B** , and it can be defined as

$$S\_est = B^T X \tag{5.6}$$

Where **B** is a orthogonal separating matrix which can be proven by

$$E(XX^T) = BE(S\_estS\_est^T)B^T = BB^T = I \tag{5.7}$$

This shows that the ICA problem to find the full rank matrix **A** can be simplified to an estimation of orthogonal matrix **B** .

Based on the Central Limit Theorem, the distribution of the sum of independent random variables tends to a Gaussian distribution. Thus, measurement of non-Gaussianity is used to find independent signals. Traditional higher-order statistics use kurtosis or named fourth-order cumulant to measure non-Gaussianity. The kurtosis of **x** is defined by

$$kurt(x) = E\{x^4\} - 3(E\{x^2\})^2 \tag{5.8}$$

For a Gaussian variance **x** , the fourth moment $E\{x^4\}$ equals to , $3(E\{x^2\})^2$ so that the kurtosis is zero for a Gaussian variable. If **x** is a non-Gaussian, its kurtosis is either positive or negative. Therefore, non-Gaussianity is measured by the absolute value of kurtosis.

In the FastICA, the kurtosis is used as a cost function to maximize the kurtosis, and negative mutual information of **X** is achieved. The FastICA is based on the fixed-point iteration scheme for finding the maximum of kurtosis [57]. The basic method of the FastICA algorithm is as follows:

1) Take a random initial vector $w(0)$ and divided by its norm. Let k = 1.

2) Let $w(k) = E\{X(w(k-1)^T X)^3\} - 3w(k-1)$

3) Divide $|w^T(k)w(k-1)|$ by its norm.

4) If $|w^T(k)w(k-1)|$ is not close enough to 1, let k = k +1, and go back to step 2. Otherwise, output the vector **w**(k) .

The FastICA algorithm needs to be performed n times to estimate n independent components, and the final vector **w**(k) should equal to the column of the orthogonal separating matrix **B** each time it is performed. Assume i **b** to be the column i in the **B** , the estimated source signal i can be expressed as

$$s\_est_i = (b_i)^T X \tag{5.9}$$

To ensure that each estimation result is a different and independent component, the FastICA algorithm adds the following simple, orthogonal zing projection

$$\begin{aligned} w(k) &= w(k) - \overline{BB^T} w(k) \\ w(k) &= w(k) / \|w(k)\| \end{aligned} \tag{5.10}$$

Where $\overline{B}$ a matrix is whose columns are the previously determined columns of **B**

The following section determines the implement of above equations in fpga

5.3 Implementation of simplified Floating-point arithmetic

To implement and design a 32 bit single precision floating point unit (FPU) for

Addition

Subtraction

Multiplication

Division

High performance FPUs are constantly needed

ASIC (Application Specific Integrated Circuits) versus FPGA (Field Programmable Gate Arrays)

ASIC – high performance but early design specifications and limited flexibility

FPGA – cost efficient and flexible

Floating Point 32 bit single precision format

S = sign bit

E = exponent

F = fraction/significand/mantissa

$(-1)^S \times F \times 2^E$ to

31	30 23	22 0
S	**Exponent**	**Significand/Fraction**

Figure 5.2 Representation of a Floating Point Number

Sign	Exponent	Significand	Value	Description
S	0xFF	0x00000000	$(-1)^S$ (infinity)	Infinity
S	0xFF	F /= 0	NaN	Not a Number
S	0x00	0x00000000	0	Zero
S	0x00	F /= 0	$(-1)^S\ 2^{-126} 0.^F$	Denormalized Number
S	0x00 < E < 0xFF	F	$(-1)^S\ 2^E\text{-}^{127} 1.^F$	Normalized Number

Table 5.1 Exceptions of a Floating Point Number

Biased Exponents

What is it?

- B = bias
- E = true exponent
- Biased exponent = 2^{E-B}

Advantages

- Easy signed magnitude representation
- Smallest exponent represented as all zeros

Implementation of Floating Point Unit is

Uses even rounded, denormalized single precision format with no bias

3 arithmetic sub-modules

- Addition/Subtraction
- Multiplication
- Division

Multiplexed output

Addition

00010011 + 00111110 = 01010001

1 1 1 1 1 carries

```
    0 0 0 1 0 0 1 1   =  19(base 10)
+   0 0 1 1 1 1 1 0   =  62(base 10)
------------------------------------
    0 1 0 1 0 0 0 1   =  81(base 10)
```

Multiplication

```
          1000
X         1001
--------------
          1000
         0000
        0000
       1000   0
--------------
       1001000
```

Division

10000111 ÷ 00000101 = 00011011

```
             1 1 0 1 1    =  27(base 10)
1 0 1 ) 1 0 0 ¹0 0 1 1 1   =  135(base 10)
      - 1 0 1             =  5(base 10)
          1 1 ¹0
        - 1 0 1
              1 1
            -   0
              1 1 1
            - 1 0 1
                1 0 1
              - 1 0 1
                    0
```

5.3.1Floating Point Adder/Subtractor

Addition and Subtraction

First: Align mantissas with largest input's exponent

Second: Adds or subtracts (via 2's complement)

Third: Normalize sum

Subtract exponent (d = e_A- e_B).

Align significands.

Shift right d positions the significand of the operand with the smallest exponent.

Select as the exponent of the result the largest exponent.

Add (subtract) significands and produce sign of result.

Normalization of result and adjust the exponent.

Determine exception flags and special values.

Basic Algorithm:

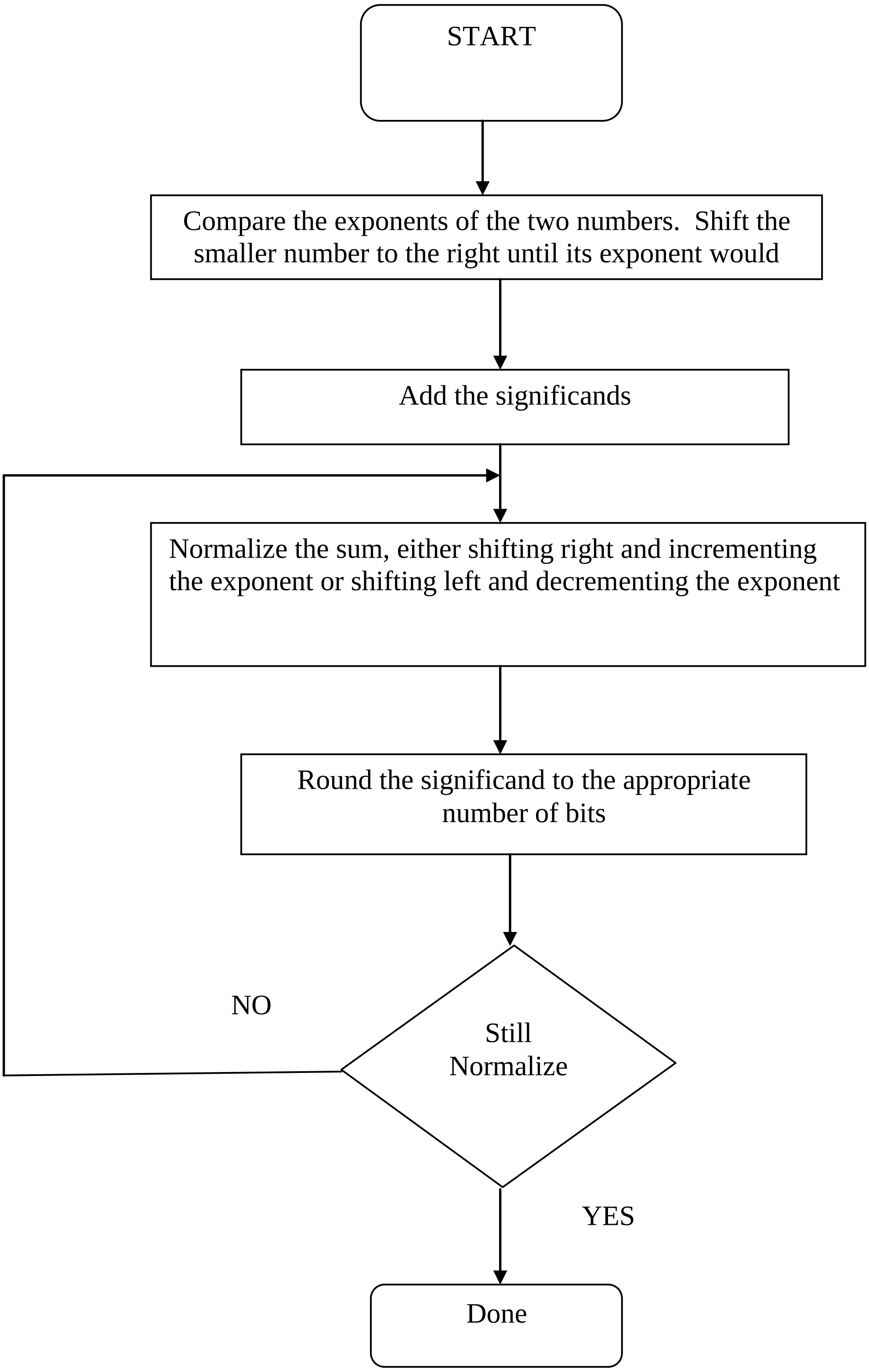

Figure 5.3 Floating Point Adder/Subtractor Algorithm Flowchart

5.3.2 Floating Point Multiplier

Multiplication

Depending on the current and previous bits, do one of the following:

00: a. Middle of a string of 0s, so no arithmetic operations.

01: b. End of a string of 1s, so add the multiplicand to the left half of the product.

10: c. beginning of a string of 1s, so subtract the multiplicand from the left half of the product.

11: d. Middle of a string of 1s, so no arithmetic operation

Shift the product register right 1 bit. Example: 2 x 6; m=2,p=6;

m = 0010

p = 0000 0110

p

0000 0110 0 no-op

0000 0011 0 >> p

1110 0011 0 p = p - m

1111 0001 1 >> p

1111 0001 1 no-op

1111 1000 1 >> p

0001 1000 1 p = p + m

0000 1100 0 >> p

=12

(a) Multiply the significands.

(b) Add the exponents.

(c) Determine the sign of the result.

(d) Normalization of result.

(e) Round.

(f) Determine the exception flags and special values

Basic Algorithm:

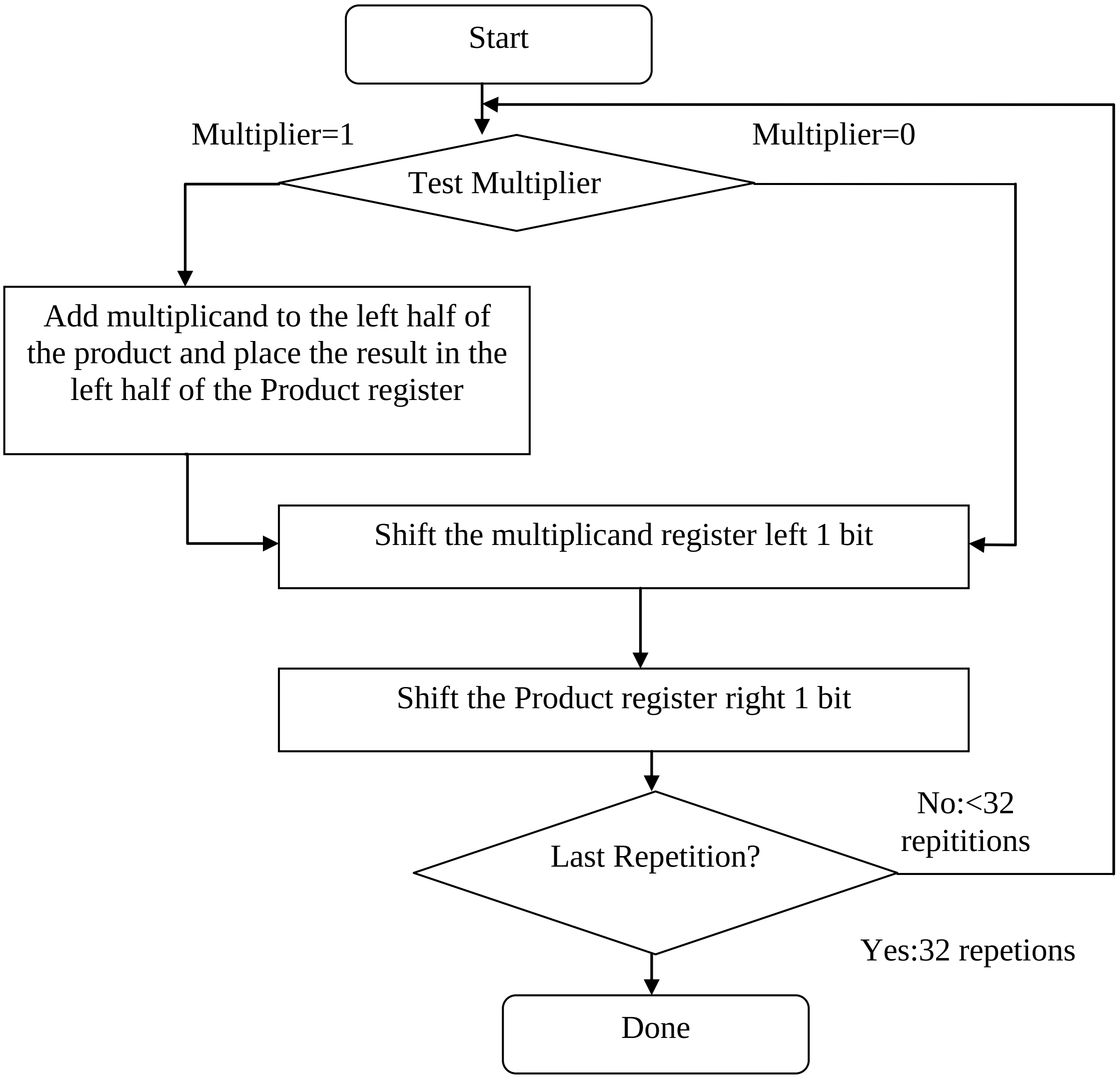

Figure 5.4 Floating Point Multiplier Algorithm Flowchart

MULTIPLICATION

First: Checks if either input is zero

Second: Shifts inputs absolute right

Third: Booth's multiplication algorithm

Fourth: Normalize product

5.3.3 Floating Point Divider

Basic Algorithm:

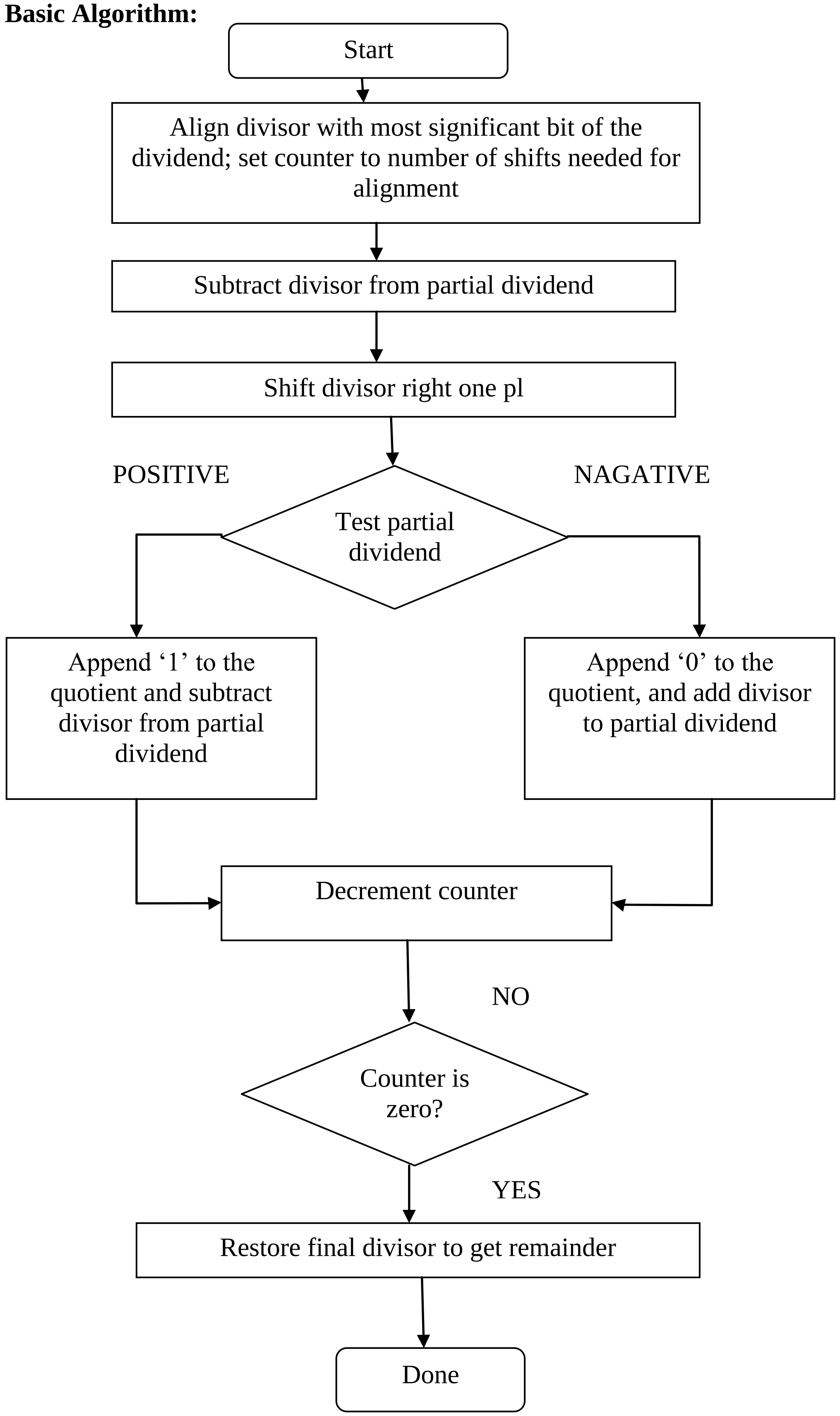

Figure 5.5.Floating Point Divider Algorithm Flowchart

Basic Algorithm:

First: Checks special cases

Divide by zero (NaN (10) or infinity(11))

Only input 1 is zero (01)

Second: Shifts inputs to the left

Third: Non-restoring division of mantissas

Fourth: Converts quotient to binary

Fifth: Normalizes

CHAPTER 6

RESULTS

6.1 Introduction

An OFDM system was modeled using MATLAB to allow various parameters of the system to be varied and tested. The aim of this simulation is to measure the performance of MIMO-OFDM with ICA equalization techniques. ICA technique was used to separate the received OFDM symbols, then the transmitted symbols are estimated and finally the Bit Error Rate curves are plotted.

6.2 SISO – OFDM

The basic OFDM transceiver is shown in Figure 2.5. The random binary data are applied to BPSK modulator. The output is a baseband representation of the modulated signal. The input must be a discrete-time binary-valued signal. If the input bit is 0 or 1, then the modulated symbol is exp(θ) *or* -exp(θ) respectively, where θ is the Phase offset parameter. This frequency domain data is then applied to IDFT block. The IDFT block computes the Inverse Fast Fourier Transform (IDFT) of length-M input, where M must be a power of ‘2’. The AWGN channel block adds white Gaussian noise to a real or complex input signal.

At the receiver basically the reverse operation to the transmitter will be done. After the removal of guard band the DFT block computes the Discrete Fourier Transform (DFT) of length-M input. The data is converted back into frequency domain so that it can be processed by the BPSK Demodulator block. This block demodulates a signal that was modulated using the binary phase shift keying method. The input is a baseband representation of the modulated signal. The input must be a discrete-time complex signal. The block maps the point *exp(*θ*)* and - exp*(*θ*)* to 0 and 1, respectively, where θ is the Phase offset parameters.

The bit error rate is calculated by comparing the input data from the transmitter with output data from the receiver. It calculates the bit error rate by dividing the total number of unequal pairs of data elements by the total number of input data elements from source as a function of Signal to Noise ratio (SNR). The same simulation work is done for QPSK and 8-PSK modulation schemes also. The obtained results are compared. The simulation results shows that BPSK outperforms remaining cases as there is less chance for error probability when compared with QPSK or 8-PSK. Hence, as the order of baseband modulation (M) increases, the probability of error also increases. The simulation results are shown in Figure 6.1.

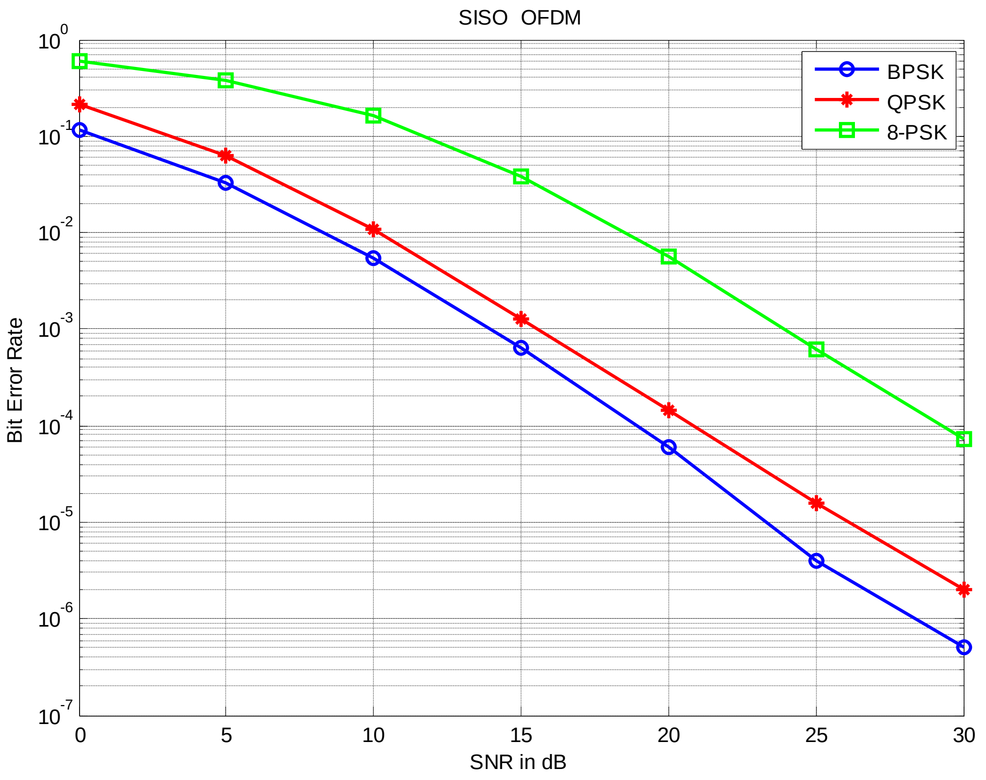

Figure 6.1: Comparison of different modulation schemes in SISO – OFDM.

6.3 MIMO – OFDM

To meet the ever growing demand for higher data rates in wireless communication systems, multiple transmit and receive antennas can be employed to make use of the spatial dimension by transmitting data in parallel streams. Such spatial multiplexing Multiple-Input Multiple-Output (MIMO) systems have been shown to obtain significantly higher data rates than Single-Input Single-Output (SISO) systems. This increase in data rate can be achieved without the need of additional bandwidth or transmit power, provided that sufficient multipath diversity is present.

Figure 5.2 compares the BER results of SISO – OFDM with that of MIMO – OFDM using one transmitting and 2 receiving antennas (1 x 2) thus providing spatial diversity. The simulation is done using 16-QAM modulation scheme, with 32 carriers and 10^4 bits transmitted. As expected, the simulation results shows that MIMO – OFDM gives better performance over SISO – OFDM for high SNR values. If we go for more number of receiving antennas, the performance still improves with increasing complexity in the receiver design.

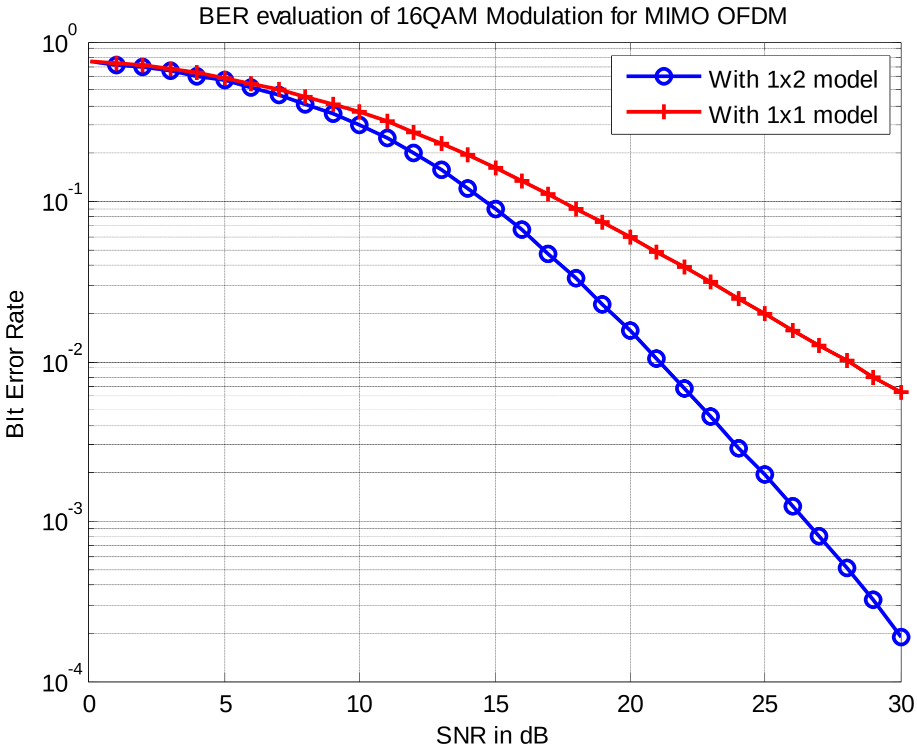

Figure 6.2: Comparison of MIMO –OFDM with SISO – OFDM.

6.4 ICA

Three independent signals are combined and separated again using ICA algorithm. These three signals are a sinusoidal wave, a square wave and a random noise-like wave, which are statistically independent. These source signals are shown in figure 6.3.

During the ICA process, a matrix is constructed from all the source signals by mixing them. This matrix is made to have zero mean for further processing. The ICA algorithm consists of 2 processes, namely “Whitening” and “Centering”.

Centering matrix does the job of making mean zero. Whitening process applies linear transformation to the resultant matrix and results in an orthogonal matrix. The separated source signals can be estimated by multiplying this orthogonal matrix’s inverse with the centered matrix. The orthogonal matrix is shown row wise in figure 6.4 and the ICA separated sources are shown in figure 6.5. On comparing the source signals (figure 6.3) with the estimated signals (figure 6.5), the results show that better estimation can be achieved provided the independency of source signals is high. Otherwise, finding the best linear transformation becomes difficult.

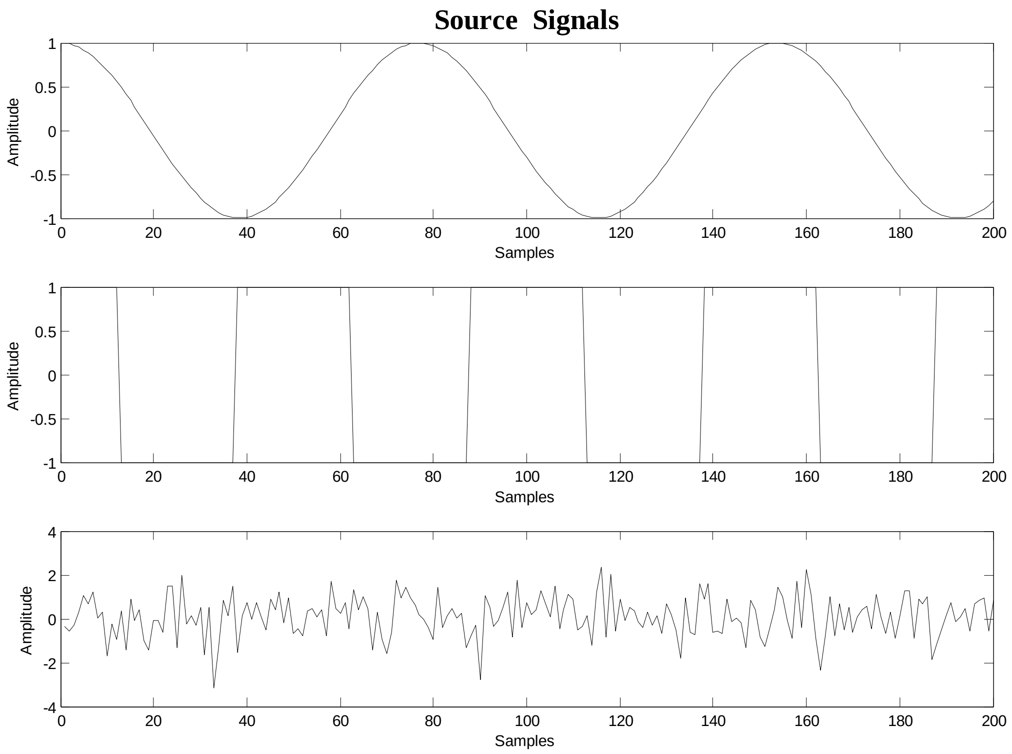

Figure 6.3: Independent source signals

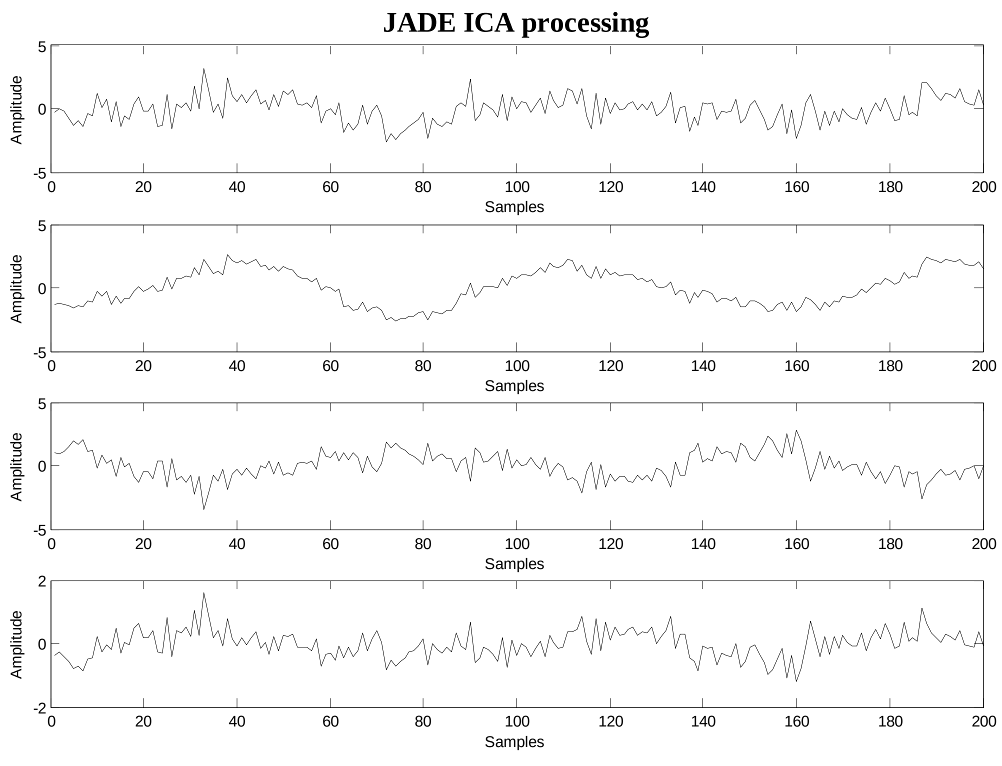

Figure 6.4: Orthogonal transformation matrix represented row-wise

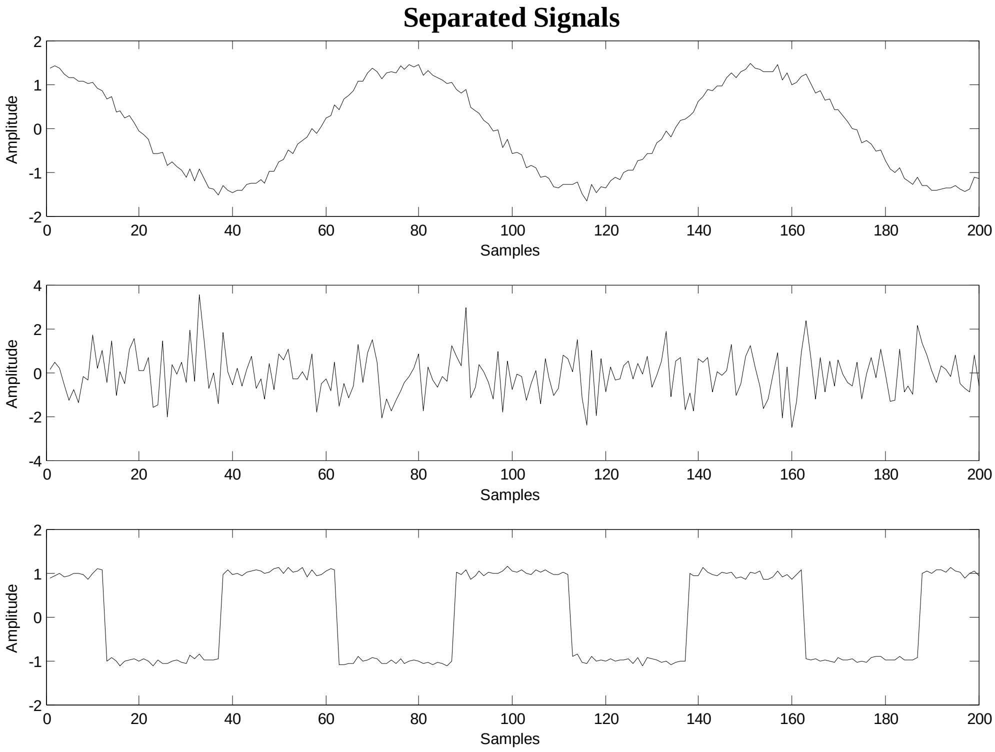

Figure 6.5: Estimated source signals after ICA processing

6.5 FPGA implementation of ICA algorithms

However, the ICA-based algorithms for convolutive mixtures with multipath reverberation require enormous computing power in real time. Also, the algorithms are memory intensive and conventional digital signal processor (DSP) architecture is not efficient. Although a few analog VLSI implementations had been reported for ICA algorithm, they were applicable to instantaneous mixtures only and were unable to be utilized for real-world speech-enhancement applications with convolutive mixtures for convolutive BSS problems, only a few VLSI architectures had been presented without actual implementation and experimental results. A DSP implementation was reported for only a simplified mixing condition with two closely spaced microphones .An FPGA implementation of digital chip is reported with modular design concept.

In Very Large-Scale Integrated Circuit (VLSI) technology has allowed designers to implement large complex designs on Application-Specific Integrated Circuits (ASICs) with millions of transistors. Field Programmable Gate Arrays (FPGAs) that is a programmable device in the ASIC family has been honored the best selection for fast design implementation.

6.5.1 FLOATINGPOINT ADDER

Fpa = 2= 010000000 00000000000000000000000

Fpb =5= 010000001 01000000000000000000000

Sum = 7= 0 10000001 11000000000000000000000

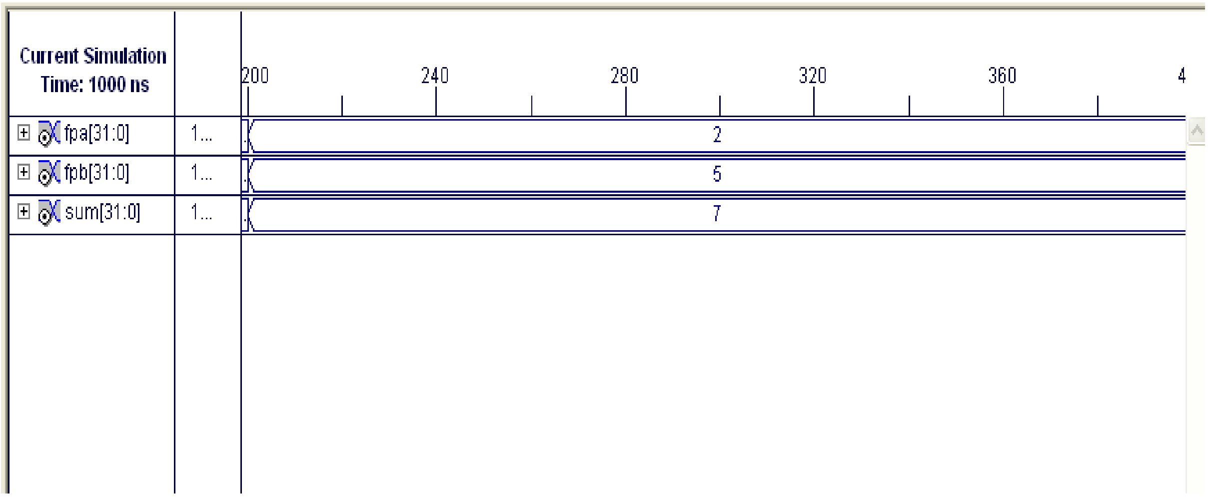

Figure.6.6 Simulation of floating-point adder

6.5.2 FLOATINGPOINT MULTIPLIER

Fpa = 2 = 0 10000000 00000000000000000000000

Fpb = 5 = 0 10000001 01000000000000000000000

Prdt_f =10= 0 10000010 01000000000000000000000

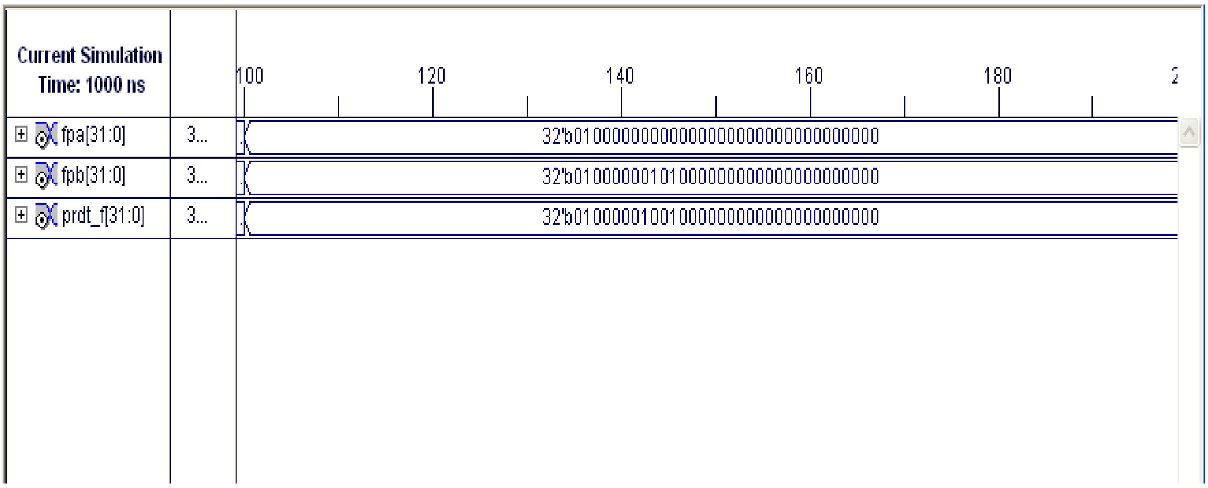

Figure.6.7 Simulation of floating-point multiplier

6.5.3 FLOATING POINT SUBTRACTOR

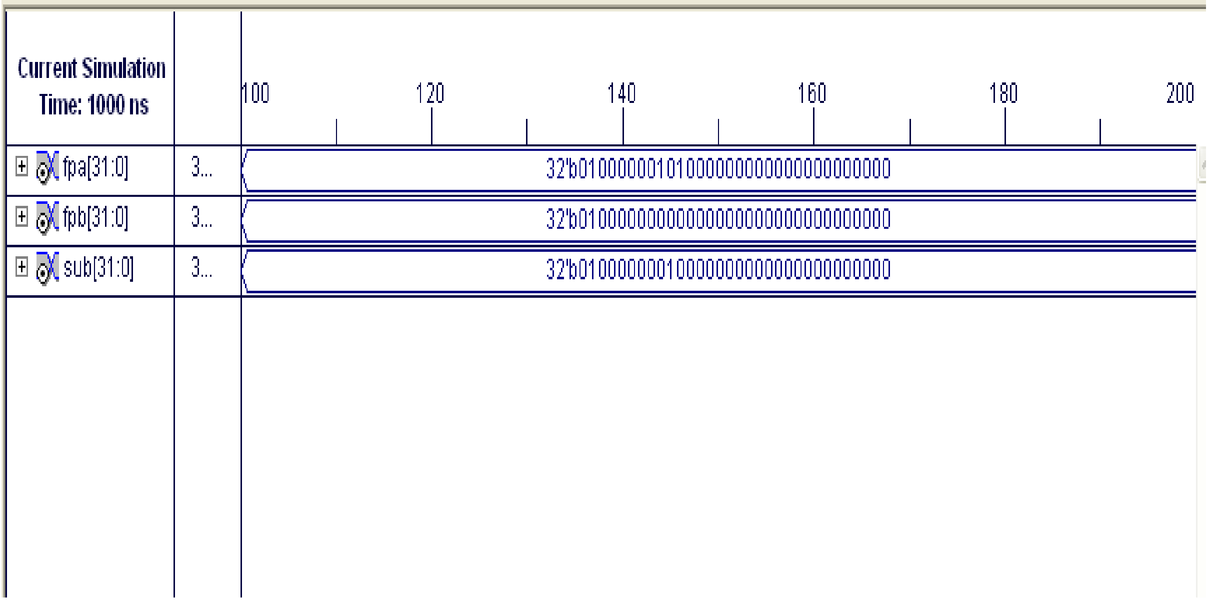

Figure.6.8 Simulation of floating-point Subtractor

6.5.4 FLOATING POINT DIVISION

A =8 = 0 10000010 00000000000000000000000

B =2 = 0 10000000 00000000000000000000000

qout=4= 0 10000001 00000000000000000000000

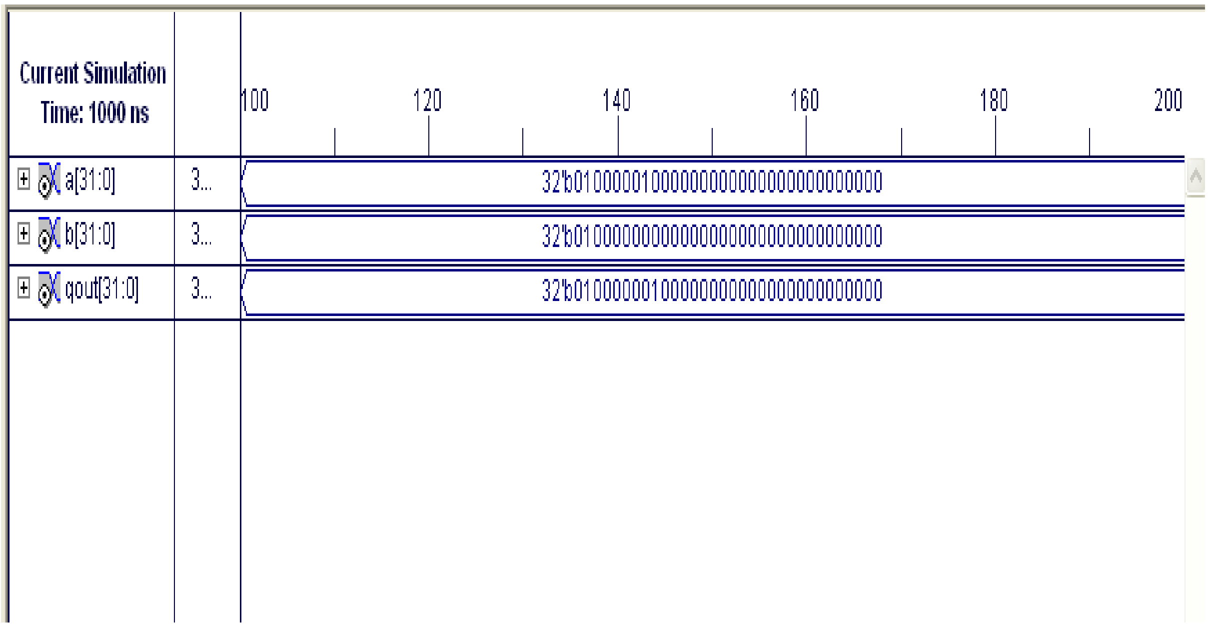

Figure.6.9 Simulation of floating-point Division

CHAPTER 7

Conclusions and future work

7.1 CONCLUSION

The MIMO-OFDM gives better BER performance over SISO – OFDM for high SNR values. MIMO channel capacity increases by a factor equal to the no. of antennas used over that of a SISO channel. MIMO system uses spatial multiplexing to increase the effective SNR of the system. The MIMO-OFDM system capacity increases with increase in diversity, i.e. no. of receivers.

The Fast ICA algorithm used in this thesis separated three different signals using blind source separation techniques. The results will be accurate, provided the source signals should be statistically independent. The main problem with Fast ICA approach is that it requires to be executed ‘n’ times, where ‘n’ is the no. of independent source signals. As the number of sources increase, the execution time increases linearly.

As the ICA technique requires HOS, its FPGA implementation needs to be in floating-point arithmetic. The basic building blocks for HOS implementation like Adder, Subtractor, Multiplier and Divisor are implemented for 32-bit arithmetic and tested with different combination of inputs. They worked well for both Integer number arithmetic and Real number arithmetic.

7.2 FUTURE WORK

MIMO-OFDM receiver can be designed using ICA equalizer. MIMO-ICA equalizer can be implemented through VLSI technology by considering FPGA floating point implemented Adder, Subtractor, Multiplier, Divider and Square root finder as the fundamental building blocks. The MIMO-OFDM can be applied to IEEE Wi-max (802.16) standards.

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